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ABSTRACT

Data on the basic components of teleconferencing and descriptions of its uses in education, business, and industry are provided in 17 individual reports. These reports include overviews and case studies of educational telephone networks, teleconferencing transmission systems and equipment, visual systems for teleconferencing, slow scan television, controlled scan television, an automatic telephone access library storage system, computer control of teleconferencing, and combining teleconferencing with tele-writing in continuing engineering education. (CMV)

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TECHNICAL DESIGN FOR AUDIO TELECONFERENCING

University of Wisconsin—Extension
Center for Interactive Instructional Programs

Compiled by
Dennis A. Gilbertson
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INTRODUCTION

The 1970's have seen a rapid growth in the non-traditional uses of telecommunications, particularly in the area of teleconferencing. Educational institutions have turned to teleconferencing as a flexible means to meet the changing needs and life styles of students. Business and industry are making increased use of teleconferencing to lower the cost of training and travel to meetings.

This means that educators and businessmen need to understand telecommunications, and that those in the telephone industry need to know how their facilities are being used by these groups. This book is a beginning attempt to combine information from both of these areas, and to meet the growing demand for this material. This publication does not pretend to be fully inclusive of all information in this field, but rather a starting point on which to build.

This first part of the book provides data on the basic components of teleconferencing, and the last section describes expanded uses.

Planning and organizing a project of this nature involves the help of too many people to list, however, there are those whom I would like to especially acknowledge. Many thanks must go to those who wrote papers and prepared their manuscripts for this publication. A special thanks to Dr. Lorne A. Parker, Director, Center for Interactive Instructional Programs (CIIP), for his advice and guidance in this effort, and thanks to Chris Olgren for her input and cooperation. Thanks also to Dean Luke F. Lamb, Educational Communications, for his continuing support of CIIP's projects. Finally, a special thank-you to the staff of CIIP and Instructional Communications Systems for their assistance.

NOTE

Some of the essays in this book have appeared in other publications of the University of Wisconsin-Extension. Articles reprinted from The Status of the Telephone in Education include: "ETN: A Technical System" and "Teleconferencing Station Equipment." The Telephone in Education, Book II contains: "Introduction to Teleconferencing," "Audio Teleconferencing: Transmission Systems and Considerations," "Controlled Scan Televideo: Efficient Utilization of Communications Channels for Teleconferencing and Data Access," and "Computer Control of Teleconferencing."

TABLE OF CONTENTS

Introduction to Teleconferencing	Lorne A. Parker, Director Marcia A. Baird, Assistant Director Dennis A. Gilbertson, Technical Director Instructional Communications Systems University of Wisconsin-Extension	1
ETN: A Technical System	Daryl Braun Vice President and Director of Engineering Darome, Inc. Harvard, Illinois Dennis Gilbertson Technical Director Instructional Communications Systems University of Wisconsin-Extension Henry C. Hansen Engineer, Transmission Systems Wisconsin Telephone Company	28
Audio Teleconferencing Transmission Systems and Considerations	R. W. Ruedisueli A. T. & T. Co. Long Lines Washington, D. C.	52
Teleconferencing Station Equipment	Daryl Braun Vice President Darome, Inc. Harvard, Illinois	68
ICS Production Services: An Overview	Richard Seifert Instructional Communications Systems University of Wisconsin-Extension	76
Visual Systems for Teleconferencing: Telewriting, Televideo and Facsimile	Christine H. Olgren Instructional Communications Systems University of Wisconsin-Extension	81
Slow Scan TV Teleconferencing	Glen Southworth Colorado Video, Inc.	109
Picture Transmission for Audio Teleconferencing	J. K. Hawkins Robot Research, Inc. San Diego, California	113
Teleconferencing as a Communications Tool	Hugh F. Gillogly National Sales Manager NEC America, Inc. Elk Grove Village, Illinois	116
The Talos Telenote System	Talos Systems, Inc. Scottsdale, Arizona	118
Electronic Blackboard: Have Chalk Will Travel	C. P. Torok Bell Telephone Laboratories, Inc. Holmdel, New Jersey	123
Slow Scan Television - An Experiment in Educational TV in the Copper Country of Michigan	Jim Zygaj Project Director	127
Controlled-Scan Televideo: Efficient Utilization of Communications Channels for Teleconferencing and Data Access	Stuart L. Meyer Northwestern University	130
An Audio-Visual Device Designed for Teleconferencing Use	Albert J. Nash President Revox Systems, Inc. Merrick, New York	141
Automatic Telephone Access Library Storage System	Rodney M. Sabick Project Director International Tapetronics Corporation	147

Computer Control of Teleconferencing	James R. Potter Director of System Operation and Development James D. St. John System Design Engineer Indiana Higher Education Telecommunication System Indianapolis, Indiana	153
Teleconferencing + Telewriting = Continuing Engineering Education in Wisconsin	Lowell B. Jackson, P.E. Lorne A. Parker, Ph.D., and Christine H. Olgren University of Wisconsin-Extension	164

INTRODUCTION TO TELECONFERENCING

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INTRODUCTION

Today's academic communities are being faced with ever changing issues that usually revolve around enrollment, funding, inflation, and general management of their particular institutions. For example, national statistics tell us that enrollments in higher education are currently stabilizing and are forecasted to slowly decrease in many institutions during the 1980's.

Furthermore, the Carnegie Commission on Higher Education, which was established in 1967 by the Carnegie Foundation for the Advancement of Teaching predicts that the 1990's will be a period of only slow growth for enrollment. Obviously this has become a major concern for higher education administrators because enrollment is the primary basis for funding. It determines government funding for the public institution sector and direct funding to the private sector in the form of tuition.

Inflation and its impact on funding and operating expenses is also a major concern within these academic communities. It has drastically raised their operating costs without correspondingly increasing their incomes. For example, from 1972 to 1974, inflation consumed two-thirds of a 29 per cent increase in the educational funds as appropriated in 46 states and when considering that fact, the forecasted inflation rates will probably continue to outstretch state funding for education.

Problems like these are being countered by many institutions through the creation and implementation of unique problem solving strategies such as competitive recruitment programs, reciprocal cooperative programs that provide for interinstitutional access to libraries, language labs, medical information and computer utilization. However, a new challenge is beginning to emerge.

Current demands on institutions of higher learning call for immediate and well planned alterations in the traditional methods and ideas for extending the facilities of these schools to remote areas.

Among the problems of providing off-campus instruction is the number of hours spent by instructors traveling from the resident campus to off-campus classes. Often an instructor will spend more time on the highway than in the classroom. Thus, coupled with limited numbers of offerings to distant students is also a restraint on implementing educational opportunities to students and adults in their home communities.

The changing needs and lifestyles of society are forcing educational institutions to seek new and flexible means of instruction against a background of budgetary stringencies.

Such searches are leading a number of institutions to discover the educational potential of the telephone. Although this invention is over 100 years old, it remains the basic instrument of the new communications technology and is supported by a highly developed intrastructure worldwide.

In its first century educators practically ignored the telephone as a teaching medium, but this is changing fast. After many small scale experiments in the sixties a second generation of educational telephone networks is now springing up in North America and Europe.

Although the telephone is no longer a startling innovation in communications, it has been only in the past few years that several technological advances have made it one of the more flexible means of instruction in distant learning education. The telephone is a way of helping those unable or unwilling to participate in traditional learning activities by bringing resource people not ordinarily available right into their community.

Teleteaching can be done in various ways and in various combinations including telelectures, teleconferences and telewriting.

Telelectures transmit and amplify for a group of students in one or several locations someone lecturing at another location. Teleconferences include two-way conversations between the instructor and students. Telewriting permits the transmission of written materials as well as voices between two or more geographical points.

Teleteaching can be done with very simple and inexpensive setups, or may take the form of telelecture or teleconferencing networks involving a central broadcast facility connected to a series of amplified telephones at a number of sites where students are located. At the broadcast location there may be broadcast microphones, taping equipment and teaching aide facilities, as well as audiovisual equipment such as slide projectors at the learning locations.

From the limited number of studies connected with the use of the telephone in distant learning, there is evidence that it is an effective means of learning and that students react favorably to the telephone as a teaching medium.

The telephone is currently being used in a number of ways for undergraduate and graduate courses, refresher courses, extension programming and continuing profesaional education:

- The University of Missouri, Columbia, teleconferencing system offers credit courses in the College of Education and the Medical School. The network is also

used for some noncredit programs and for administrative conferences.

- The University of Minnesota, St. Paul, uses the network for Agricultural Extension Service educational programs in such areas as horticulture, home economics, 4-H youth, and agriculture.

- At the University of Maine, Portland-Gorham, the Bureau of Continuing Education for Nursing offers courses to about 1,200 RN's and LPN's each semester.

DEFINITION OF TERMS

Since many terms used in studies to describe telephone teaching are not uniform, here are a few definitions:

Teleconferencing is the generic term covering distant two-way telephone discussion between two or more groups.

Conference calls are the type of teleconferencing in which ordinary telephone lines are linked together by a telephone operator or a connecting device. Participants could be using amplified telephones at each location or a regular telephone.

Telelecture can best be described as a prearranged telephone call from the classroom to a resource person providing students the opportunity to ask questions and to make comments. Responses are amplified through special telephone equipment for the entire classroom. Microphones and speakers may be placed in appropriate locations in the room, making it possible for all participants to more conveniently interact.

A dedicated system is a teleconferencing network in which permanently installed lines are leased from the telephone company 24 hours a day. All listening centers on a dedicated system are permanent and are wired to operate like a giant party line.

Telewriting has become synonymous with Victor Comptometer Company's electrowriter system and more recently the Talos writing systems. Telewriting is a term coined for a system that allows the transmission of drawings and graphic materials from one location to another. The graphic equipment has the ability to transmit and receive electronically written images with the use of electronic writing equipment and receivers. The image transmitted from the sending location is projected on a screen using an overhead projector at the various receiving locations. Telewriting is usually used as a supplement to teleconferencing.

The electronic blackboard system has recently been developed by Bell Telephone Laboratories and will be offered as a standard AT&T product by July 1977. To transmit normal classroom blackboard graphics or drawings, the instructor simply uses standard chalk on a regular 4' x 8' electronically sensitive standard surfaced blackboard. The handwriting motions are electronically converted and transmitted over a standard telephone line to the receiving end which can be across town or across the country. The signals are then reconverted and displayed on a television screen. The two-way interactive audio is transmitted on another telephone line and both systems can be recorded on a standard stereo tape recorder so the audio and graphics material can be played back at different class meetings at various times and locations.

WHAT IS TELECONFERENCING?

Teleconferencing is the means of joining two or more groups in different locations through a telephone system. Voice amplifiers, speakers and microphones allow remote locations to talk with the instructor and each other.

In a teleconference, phone lines transmit a speaker's voice to communities miles away. There, his words are amplified using special telephone equipment and any participant can respond if appropriate. For instance, an instructor located anywhere in the country can give a lecture simultaneously to several groups of students in different places. The instructor's voice is amplified in each classroom, and the students communicate with the teacher and other students through a microphone.

Teleconferencing can accommodate large groups located at a few different places, or small groups, even individuals, at a multitude of locations. Take, for example, a university with campuses scattered throughout the state. Equip the school with a full teleconference system. On Monday morning, administrators at each campus, no matter how far away, can meet with the central campus via teleconference. Midmorning, faculties can hold monthly meetings over the same system. Later in the day a course in fourth semester Greek could be taught from the main campus, where the only Greek professor is located, to two outlying campuses and four community colleges. That evening, an extension course in real estate law could be brought to a number of outlying community "classrooms."

Frequently it is advisable to prepare and disseminate handouts, charts, slides or other graphics to enhance the teaching-learning process.

A typical use of teleconferencing has been to conduct meetings between a number of people who need to discuss matters concerning their work. On rare occasions, these conferences take on major proportions. Nurses at the University of Quebec recently held a successful meeting with 1,754 participants in eight locations across the province.

Teleconferencing systems are used for many different educational and administrative uses including a variety of specialized programs in medicine, art, music and engineering. Existing systems range from a few sites interconnected by a telephone operator, to complex setups with 200 or more locations integrated with slides, real-time graphics, computers and video tapes.

Types of Educational Teleconferencing

There are basically two types of telephone-based educational delivery systems; the conference call system and the dedicated system.

The Conference Call System

Conference calls, normally made on an operator controlled system, simultaneously link together ordinary telephone lines at a number of locations. These calls are scheduled in advance with an operator who dials all the participants and connects them to what is known as a conference bridge. This is simply an electronic device at the telephone company's central office which allows calls to be joined together.

The conference call system has two major benefits over a dedicated system: (1) flexibility of program and location needs, (2) costs--only pay for time used.

Remember, we are still talking about amplified telephone equipment and microphones at each one of the stations on the conference call. Thus, large numbers of people would be able to hear the speaker and talk with others via the microphones.

The Army's Use of Conference Calls for Education

The U.S. Army uses the dial conference capacity of AUTOVON, the Automatic Voice Network, to conduct training from Army schools to reserves throughout the United States. In one such training program, the Army established a dial-up network connecting seven reserve units. In this way, 50 students at different locations participated in a radio teletype procedures course presented from Fort Gordon, Georgia.

Dedicated System

Unlike conference calls, which employ regular telephone service, a dedicated system uses permanently installed facilities leased from the telephone company on a 24-hour-a-day basis. Since the listening centers are permanent, all the points on the system are wired to operate like a "party line."

The University of Wisconsin's Use of a Dedicated System for Education

The University of Wisconsin-Extension has the largest telephone-based instruction system in the country. A brief explanation of this system will serve as an illustration of this type of teleconferencing.

Wisconsin's Educational Telephone Network (ETN) was developed by the University of Wisconsin-Extension in 1965 as a means of expanding educational opportunities at a minimal cost to state residents. ETN is a private four-wire telephone network that takes the form of a huge party line with a number of people on the same line.

The ETN system was initiated to help doctors throughout the state of Wisconsin keep up with medical advances. From the original postgraduate medical update, ETN has expanded not only the number of listening locations, but in the diversification of its programs. Presentations range from the medical courses to pharmacy, English, law, social work, home economics, library science, education, nursing, physics, agriculture and engineering. In addition, 4-H programs, Extension faculty conferences and weekly departmental staff meetings are carried via this system. Programs are essentially noncredit continuing education for professional adults in the state.

COMMONALITIES OF SUCCESSFUL SYSTEMS

We have explained and described the two modes of teleconferencing, the conference call and dedicated system. Now it seems worthwhile to examine several telephone systems operating today.

Wisconsin	Ohio
Florida	Oklahoma
Los Angeles	Open University
Kansas	University of Quebec
Illinois	U.S. Army

Projects presented here are growing and successful, in part, because of several factors they share. A look at what they have in common may benefit others considering teleconferencing systems.

Institutional Commitment

Although they come in all sizes, each institution described here believes the telephone is an important tool of instruction and supports it accordingly. For more than a decade, for example, the University of Wisconsin-Extension, has supported its Educational Telephone Network (ETN) as a prized and integral part of statewide outreach. It is one of a handful of communications systems used to implement the Wisconsin Idea in education . . . the idea that the boundaries of the campus are the boundaries of the state.

Types of Programs and Teaching Styles

Successful telephone-based instruction programs consist of more than merely placing a microphone in a traditional face-to-face classroom situation and sending the spoken material out over telephone lines. Instructors who have successfully developed telephone-based courses employ a number of teaching strategies designed specifically for this largely interactive audio medium. Such strategies include short 12 to 15 minute learning segments, interviewing people at different locations, reactor panels, local buzz groups, feedback assignments, site-to-site discussions, and relating questions to student occupations.

Meeting Student and Teacher Needs

Teleconferencing as an alternative educational method prospers in areas where it meets very real, pragmatic professional and economic needs. The lawyer who continually needs to keep abreast of changes in the law in order to maintain his license; the doctor whose constant need to continue his education is juxtaposed against his need to stay near the community he serves; the librarian who has 150 miles and a harsh winter climate separating her and the nearest degree granting campus . . . and the instructors who, because of professional commitments and the sheer logistics of limited time and economics, cannot serve a statewide clientele in traditional face-to-face methods.

Technical Flexibility

Teleconferencing systems need to be flexible. Within the limits of individual systems, whether using the conference call mode or dedicated networks, the system must be adaptable to varying numbers of stations. Equipment must be standardized, easy to operate and suitable for different audience sizes at individual locations. Those systems which have achieved a near maximum level of sophistication, without sacrificing flexibility or standardization are the most successful.

TELECONFERENCING TODAY

The following examples of teleconferencing represent current activity in the field.

Wisconsin

The University of Wisconsin-Extension Educational Telephone Network (ETN) is currently the largest, most sophisticated telephone-based educational delivery system in the United States.

It started in 1965 with 18 listening sites around the state. The listeners might hear the lecture quite clearly one minute, then hear nothing but static the next.

While it was up to the telephone company to work out the technical bugs, it was up to the ETN staff headquartered in Madison and the UW-Extension faculty to work out the program details.

The first program was a seminar for physicians, a group that needed classes in order to learn the latest medical knowledge, but couldn't afford the time to travel far from their patients.

This first class was a hit, and ETN was on its way. The next year three new health programs were scheduled, and the decision was made to start programs for other professionals.

Lawyers, for instance, attended ETN noon lectures while they ate their sack lunches. Over the years, they have had one of the highest turnouts of any groups--about 500 per program.

Soon programs were offered for the general public as well as professionals. These programs included law for the layman, a course on venereal disease, a discussion of the effect of TV violence on children, and another on how to be a better homemaker. Most classes are noncredit, although some credit courses are offered each semester.

In recent years, ETN classes have even been offered in photography and music. In Photo Fundamentals, for example, students mailed their class assignments to their instructor who critiqued and returned them.

Music was harder to adapt to the system, since it requires a higher level of sound quality than a lecture. The test came in the spring of 1973 with a forum for piano teachers. Artists from as far away as Texas and New York performed--via the telephone lines--and listeners were surprised at the almost concert hall quality of the sounds. Teachers demanded an encore and the program has been repeated the last three years.

Today, as the Wisconsin ETN celebrates its eleventh anniversary, there are over 100 UW-Extension courses offered each year to more than 30,000 students. A map identifying ETN community locations within each Wisconsin county is on the following page. Each of the 200 individual locations is equipped with Darome Edu-Com speakers and four microphones connected to a standard telephone facility.

SEEN--A new dimension was added to the ETN idea in 1969, when images could be transmitted along with the telephone lecture. In fact, an additional system was designed and called SEEN, or the Statewide Extension Education Network. With SEEN, the images appear on a screen at the listening site at the same time that the lecturer draws them on an electrowriter. SEEN has made it possible for UWEX faculty to teach calculus, for example, because they can now send out instant equations and formulas. So far, 23 cities have the capability to receive SEEN programs. SEEN locations use Darome Edu-Com units for audio transmission and reception and Victor electrowriter units.

Florida's ETN

For several years the College of Pharmacy has been interested in initiating an educational telephone network in Florida. A foreign graduate pharmacist's program presented the opportunity to develop the network.

In 1973 the Florida legislature mandated that foreign professionals residing in Florida be provided one year of continuing education to prepare them to take the Florida state licensing exam for their profession. The law also specified that the continuing education could be provided in their native tongue at the request of at least 15 students.

The University of Florida School of Pharmacy, Gainesville, was chosen as the agency to provide the necessary education for pharmacists. At the time the act was passed, over 600 graduates of foreign pharmacy schools were living in Florida, mostly Spanish speaking Cuban exiles. They requested that the course be offered in Spanish.

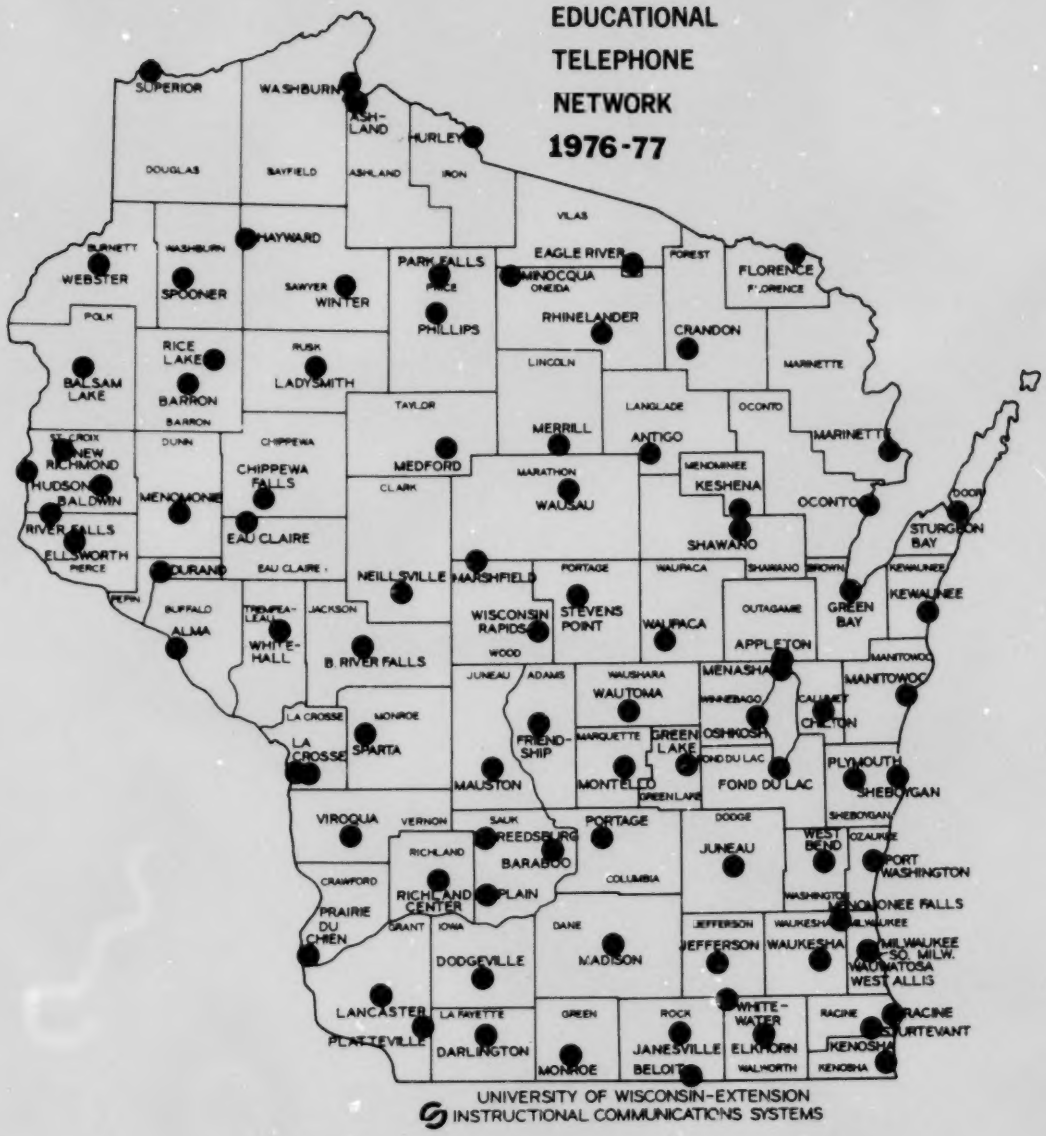
Curriculum development of the proposed courses took place in 1974-75, and methods were established for interfacing an English speaking faculty with a Spanish speaking student group. By the end of 1976 the first one year course was to be completed.

A standard Bell System multi-point dedicated voice grade facility was installed by Southern Bell Telephone Company and Darome Edu-Com station equipment was used because of the multiple stations requirement and the provision for synchronized slide projector capability.

Listening posts were established in Orlando, West Palm Beach, Tampa, Miami, Hollywood and Jacksonville.

Plans are also underway to utilize the network for other mandated continuing education requirements. The College of Education is investigating the potential of ETN for credit courses.

Control of the University Florida system is within the Division of Continuing Education.


EDUCATIONAL
TELEPHONE
NETWORK
1976-77
SUPERIOR
WASHBURN
ASH-
LAND
HURLEY
DOUGLAS
BAYFIELD
ASHLAND
IRON
VILAS
BURNETT
WASHBURN
HAYWARD
EAGLE RIVER
FOREST
FLORENCE
WEBSTER
SPOONER
SAWYER
WINTER
PARK FALLS
PRICE
MINOCQUA
ONEIDA
PHILLIPS
RHINELANDER
CRANDON
POLK
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MARINETTE
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BARRON
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TAYLOR
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OCONTO
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MARINETTE
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CHIPPEWA
MENOMINEE
NEW
RICHMOND
CHIPPEWA
FALLS
MEDFORD
MARATHON
KESHENA
HUDSON
BALDWIN
MENOMONIE
CLARK
WAUSAU
OCONTO
DOOR
STURGEON
BAY
RIVER FALLS
EAU CLAIRE
SHAWANO
ELLSWORTH
PIERCE
DURAND
EAU CLAIRE
NEILLSVILLE
MARSHFIELD
PORTAGE
WAUPACA
SHAWANO
BROWN
KEWAUNEE
KEWAUNEE
PEPIN
BUFFALO
TREMPEA-
LEAU
JACKSON
WISCONSIN
RAPIDS
STEVENS
POINT
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BAY
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WHITE-
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WOOD
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ADAMS
WAUSHARA
MENASHA
WAUTOMA
WINNEBAGO
CALUMET
MANITOWOC
CHILTON
LA CROSSE
MONROE
OSHKOSH
FRIEND-
SHIP
MARQUETTE
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SPARTA
FOND DU LAC
PLYMOUTH
MAUSTON
MONTELLO
GREEN LAKE
SHEBOYGAN
VIROQUA
VERNON
SAUK
PORTAGE
DODGE
SHEBOYGAN
REEDSBURG
JUNEAU
WEST
BEND
OZAUKEE
RICHLAND
BARABOO
PORT
WASHINGTON
CRAWFORD
COLUMBIA
WASHINGTON
RICHLAND
CENTER
PLAIN
MENOMONEE FALLS
PRAIRIE
DU
CHIEN
GRANT
IOWA
DANE
JEFFERSON
WAUKESHA
MILWAUKEE
JEFFERSON
WAUKESHA
MILWAUKEE
SO. MILW.
WAUWATOSA
WEST ALLIS
DODGEVILLE
MADISON
LANCASTER
LAFAYETTE
GREEN
ROCK
WHITE-
WATER
RACINE
RACINE
STURTEVANT
PLATTEVILLE
DARLINGTON
JANESVILLE
ELKHORN
KENOSHA
MONROE
BELOIT
WALWORTH
KENOSHA
UNIVERSITY OF WISCONSIN-EXTENSION
INSTRUCTIONAL COMMUNICATIONS SYSTEMS

Los Angeles Dial-A-Class

The dial-a-class concept has been around for a long time but with recent developments in technology it has been perfected to an effective educational communications channel. The Belmont Community Adult School, Los Angeles, has been using this system to teach over 500 adult students receiving classroom instruction via the telephone in 25 subjects throughout their district.

The dial-a-class program allows the school district to provide continuing education to anyone who desires to learn. The program started a few years ago to provide students classes of their choice in their homes, in convalescent homes, while in hospitals or any place where they have access to a telephone.

No additional equipment is needed at the listening end unless a headset type receiver or speaker is desired for the system. Teachers contact the classes from a central telephone instructional center. The teacher dials the student's telephone number using a dial card and students are immediately piped into the class. The system has the capability of connecting 18 locations.

At the present time Belmont Community Adult School system students can earn high school credits and take classes on a noncredit basis. Classes offered include United States history, California history, English as a second language, world literature, music masters appreciation, conversational Spanish, mathematics review and conversational English.

Here's how the dial-a-class works:

1. The teacher calls each student to the telephone class with an automatic dialer, quickly and accurately. The teacher may use a speaker phone or operator type headset: both allow hands-free convenience.

2. The student answers the ring of their home telephone, specially provided by the school, and are connected into the teacher's console on a conference hook-up. As a regular classroom, they can chat together while waiting for the rest of the class to arrive on the telephone.

3. After each student is connected, the teacher places his or her card into a special slot on the card dialer console. The student's name is visible and a lamp by each slot flickers when he speaks. The teacher can see at all times who is talking or making noises.

4. As the class proceeds, all students hear the teacher's questions and all hear the answer given by the student called upon.

Features of the dial-a-class are:

1. A teacher can operate the push-button console.

2. A teacher can isolate any student's line to speak with him or her in private.

3. Students can be separated from hearing each other, while the instructor hears and talks with them all.

4. The class can be divided with some students participating in discussion while others work alone on an assignment.

5. Up to nine students can be transferred to another teacher in another location.

6. Recorded teaching materials can be presented over the system.

Kansas

The Regents' Continuing Education Network, installed in 1970, links classrooms in 26 Kansas communities by private four-wire telephone lines. Six Kansas colleges sponsor and develop network credit and noncredit courses. Participating schools include Fort Hays State College, Kansas State College of Pittsburg, Emporia Kansas State College, Kansas State University, Wichita State University and the University of Kansas.

In addition to the above campuses, telephone network classrooms are also located in such places as community junior colleges, vocational-technical schools, grade schools and adult education centers. Each classroom uses a Western Electric 50-A Portable Conference Telephone to participate in the programs.

The Kansas network operates approximately 40 hours each week and involves three types of programs: credit programs, such as Education of Exceptional Children, Problems in Business Administration and Basic Nutrition; noncredit continuing education programs; and in-service training programs contracted by small businesses.

Illinois TeleNet

Illinois TeleNet, Urbana, Illinois, is a 48 station dedicated telephone network of three subsystems which can be used independently or joined together.

One network is a dedicated four-wire system connecting each of eight regional Extension offices, the Extension directors office and two other administrative offices. This network is used for management of state Extension service.

A second network connects all of the offices in Extension region one in the northwest corner of the state and the regional director's office for regional and intercounty communication meetings.

Finally, a 31 station network links other county Extension offices throughout the state for intercounty communication, agent updates and general public programming.

During 1975-76, 640 instructional hours of programming were offered to an estimated 33,869 persons. TeleNet programs have included a tax course, a workshop on how to make visuals, home landscaping, severe weather, legislative updates, interior design and consumer education.

Equipment used for TeleNet is the Western Electric 50-A Portable Conference Telephone. Currently the Western Electric 50A-1, an improved model of the 50-A, and a unit made by Darome, Inc., are being tested on the network. Counties without dedicated TeleNet

installations can also be connected to a program via a standard telephone facility and a Portable Conference Telephone.

Illinois is exploring the feasibility of every county office having its own TeleNet unit.

Ohio OMEN (Ohio Medical Education Network)

OMEN, Columbus, Ohio, is a two-way audio network linking 95 hospitals in seven states and Canada. It bills itself as the "largest two-way physician's network in North America" and is currently in its 15th year of operation.

Each week the network features discussions between hospital staff physicians and medical school faculty. The discussions link community physicians at hospitals in Ohio, Kentucky, Indiana, Illinois, West Virginia, Pennsylvania, Michigan and Ontario with faculty members at four medical schools: Case Western Reserve University at Cleveland, University of Cincinnati, Medical College of Ohio at Toledo and Ohio State University College of Medicine, Columbus.

During 1975-76, 20,500 people registered in the 125 programs offered. The network is used about 140 hours each year.

To participate in OMEN, each hospital needs two telephones. A regular telephone is used to call in questions and comments to Columbus during the programs. A second telephone receives the program. The receiving line is hooked to network-specified amplifier and loudspeaker equipment at each hospital.

Oklahoma Regional Medical Program

The Oklahoma Regional Medical Program, Oklahoma City, is a dedicated network linking more than 90 hospitals. Among the locations are hospitals in Oklahoma, six in Kansas, and one in Texas. The teleconference network is headquartered within the University of Oklahoma Health Sciences Center.

About 25 courses are offered on a regular basis to supplement hospital in-service education programs. The network is particularly aimed at serving the continuing education needs of doctors and allied health professionals in rural areas.

The Oklahoma program uses standard Bell System dedicated facilities and Darome Edu-Com units for station equipment.

The Open University-London, England

In 1973, this English correspondence institution, which has an enrollment of some 58,000 students, began experimenting with teaching by telephone. While the correspondence unit remains the heart of the Open University, radio, television, face-to-face tutoring and educational counseling at the system's 260 study centers enrich the program. Experiments in teleconferencing have arisen largely because of problems faced by the widely scattered student population and their difficulties in attending the study centers.

In 1975, some 171 students participated in teleconferencing through the Open University. These students formed 25 different groups which studied a variety of courses in math, science, technology and the arts. Individuals were contacted by telephone at their homes and met for about an hour every two weeks. This program was quite successful and experiments are now underway with loudspeaking telephone equipment at the study centers to ascertain the viability of group tutoring and teaching.

University of Quebec

The University of Quebec has used teleconferences as a routine administrative tool for several years. Founded in 1969, the University of Quebec enrolled 30,000 students in 1974-75. Its 20 campuses, centers and research institutes are found in eight cities over an 800 mile area. Thus, the installation of a teleconferencing system helped administrators reduce travel among the various institutions. The University is currently renting about 20 Western Electric Portable Conference Telephone units.

For University of Quebec administrators, the average teleconference involves seven people and lasts about an hour and fifteen minutes. In a two year period, over 25 university committees, involving some 150 people, used this system.

United States Army

In 1975, the United States Army experimented with teleconferencing for in-service reserve training. Using the dial conference capacity of the Army's existing Automatic Voice Network, AUTOVON, a seven-point conference call network was established.

The pilot program linked approximately 50 reserves studying a radio teletype procedures course with instructors from Fort Gordon, Georgia.

Each location was equipped with a Darome two-wire Convener which provided two-way communication between sites, and featured the capacity for remote control of a Kodak carousel.

The Army currently is using teleconferencing to teach medical subjects, officer career courses and helicopter maintenance.

PERSPECTIVE: A BRIEF HISTORY

We have examined educational telephone systems of today, but let us know take a look back to the beginning and see how teleconferencing developed.

Experiments in the use of the telephone for all levels of education, from elementary to graduate, have been going on for nearly three decades. The following brief highlights of where teleconferencing has been will give insight into its future possibilities.

One of the earliest uses of the telephone for education was in 1939, when it was employed to meet the needs of homebound and hospitalized students in Iowa. Within two years, more than 1,000 students were receiving class lectures through intercom equipment in their homes and hospital rooms, at the same time the lessons were being taught to classroom students. All were able to talk with their teachers and fellow students.

Teleconferencing at the college level began in

1947, at the College of Dentistry of the University of Illinois Chicago Medical Campus. Dentists in Scranton, Pennsylvania, took part along with dentists in the Chicago campus classroom.

Of the many pilot amplified telephone programs initiated in the 1950's, the most extensive was that of Stephens College, Columbia, Missouri. Begun in 1958, it was first used for in-service training of science faculty at several colleges. Later, several area colleges and universities employed the system for faculty and facility sharing. Also, through this teleconferencing system, such speakers as Sen. Robert Taft and Harold Stassen were brought to the campus.

In 1960 the University of Omaha began a telelecture program, reaching audiences of up to 500. Later they joined with the University of Colorado in a "Share the Professor" telelecture plan to exchange undergraduate credit classes three times each week.

Wisconsin's Educational Telephone Network (ETN), which has been described earlier, began five years after Omaha's first program.

In 1966, the University of West Virginia started their telelecture program for in-service training to isolated Appalachian school administrators.

Until that year, remote lectures could not be supplemented with the creation of written material as in the classroom. Cornell University's College of Engineering, however, combined telephone and television technologies to teach a course in physical metallurgy to research specialists located 55 miles away in Pennsylvania. The system transmitted both the oral lecture and handwritten materials over telephone lines.

Electrowriters have proven to be an important accessory to teleconferencing. Victor Comptometer Company, Chicago, Illinois, generally is credited with the development and marketing of electrowriters. Wisconsin and West Virginia are just two of the states using this equipment today.

Other telephone networks continued to spring up throughout the Midwest and West during the latter half of the 1960's. Networks in Illinois, Missouri, Kansas, Oregon, Nevada, Wyoming and Minnesota were formed.

The Canadian Department of Indian Affairs also began teleconferencing for administrative purposes in 1972.

LOOKING AHEAD - ELECTRONIC BLACKBOARD

A telewriting feature recently developed by Bell Telephone Laboratories is a system that transmits blackboard writing over ordinary telephone lines for display on a television screen at distant locations.

The electronic blackboard is far less costly than closed circuit television, the only comparable system transmitting both audio and visual information instantaneously. The new device uses ordinary telephone lines and does not require professionally trained technicians to operate.

The University of Illinois, being the trial University, has used the electronic blackboard to successfully conduct remote engineering classes from its campus in Champaign-Urbana to off-campus locations in Freeport, Rock Island, Peoria and Rockford. The system has recently been approved to be introduced into the Bell System as a standard product offering for utilization throughout the United States.

To transmit hardwriting or other graphics, the instructor simply writes on a standard 4' x 8' blackboard which is electronically sensitized such that with normal chalk, remote receivers can view them via standard video monitors on a "real-time" basis.

The interactive audio portion of the lecture is carried over a second telephone line, via a portable conference telephone already offered in many forms by the Bell System.

An interactive teleconferencing system with real-time graphics will increase the capability of remote education and both technical and administrative conferencing. Since the system transmits in real time, classes can be conducted live; however the system is designed to enable an entire lecture including both the interactive audio and graphics to be recorded on a standard stereo tape recorder for subsequent retrieval. For example, if students or conferees are unable to attend the original lecture or conference, they can come to the remote location and replay the tape at their convenience. In this instance, the system replays both the audio and graphics.

THE ECONOMICS OF TELECONFERENCING: AN ILLUSTRATION

Interest in the development of cost per student contact hour of instruction in both face-to-face and media situations has risen to new levels during the past decade. Concern for increased accountability is a dominant theme in a time of economic retrenchment.

The purpose here will be to explore one example of the economics of dedicated facility interactive audio systems, audio graphic systems and video systems as compared to face-to-face instruction.

The base systems cost data used here is from a study conducted by Ted Augustine, former AT&T Marketing Supervisor-Education, on cost comparisons of tying three compuses of Pace University (Manhatten, Westchester, White Plains) and, the Long Island Technical Institute, New York, together via audio-graphic and video systems. Each system will be examined separately.

Base data used in this study includes instructor salary, student contact hours, equipment and facilities.

Instructor Salary

Computations here are based on an annual instructor salary of $18,700. A full teaching load consists of three 3 credit courses each semester.

$18,700 ÷ 6 courses/yr = $3,116.66 salary for one course.

Student Contact Hours (SCH)

SCH = number of students x total course hours.

For example, a 15 week course that meets three hours each week with 20 enrollees = 15 x 3 x 20 = 900 SCH.

Equipment, Facilities, and Personnel

For telephone-based systems, equipment and facilities include control studio for bridging, tape inserts, tape cueing, prerecorded programs, volume control, etc. Also included are program director, classroom studio, terminal equipment and remote classroom with telephone access jacks.

Number of Students

Computations are based on a class size of 20 students for face-to-face instruction. Teleconferencing stations at the four campuses are also assumed here to attract 20 students each or a total of 80 students.

EXAMPLE 1 -- FACE -TO-FACE

Face-to-Face
-3 credit course to 20 students

Student Contact Hours (SCH)
-20 students x 3 Hrs/Wk x 15 Wks = 900 SCH

Cost
-Salary for one 3 credit course = $3,116.66

Cost Per Student Contact Hour
-$3,116.66 ÷ 900 = $3.46/SCH

EXAMPLE 2 -- AUDIO

Audio to four campuses
-3 credit course to 80 students

Student Contact Hours (SCH)
-80 students x 3 Hrs/Wk x 15 Wks = 3600 SCH

Cost
-System cost of $1,250/Mo
-$312.50 Wk ÷ 3 Hrs/Wk = $104.17/Hr
-45 Hr Course x $104.17 = $4,687.49
-Salary for one 3 credit course = $3,116.66
-$3,116.66 + $4,687.49 = $7,804.15

Cost Per Student Contact Hour
-$7,804.15 ÷ 3600 = $2.16/SCH

EXAMPLE 3 - AUDIO GRAPHIC

Audio Graphic to four campuses
-3 credit course to 80 students

Student Contact Hours (SCH)
-80 students x 3 Hrs/Wk x 15 Wks = 3600 SCH

Cost
-System cost of $3,200
-$800/Wk ÷ 3 Hrs/Wk = $267.00
-45 Hr Course x $2.67/Hr = $12,015
-Salary for one 3 credit course = $3,116.66
-$3,116.66 + $12,015 = $15,131.66

Cost Per Student Contact Hour
-$15,131.66 ÷ 3600 = $4.20/SCH

EXAMPLE 4 -- VIDEO

Video to four campuses
-3 credit course to 80 students

Student Contact Hours
-80 students x 3 Hrs/Wk x 15 Wks = 3600 SCH

Cost
-System Cost of $24,000/Mo
-$6000/Wk ÷ 3 Hrs/Wk = $2,000.00
-45 Hr Course x $2,000/Hr = $90,000
-Salary for one 3 credit course = $3,116.66
-$3,166.66 + $90,000 = $93,116.66

Cost Per Student Contact Hour
-$93,116.66 ÷ 3600 = $25.86/SCH

However, if we begin adding classes onto any of the three technological distribution systems, since our basic costs are fixed, the price per student contact hour decreases quite rapidly.

For example, the hourly cost for an audio system used 3 hours each week is $104.17 an hour, or $2.16 per student contact hour for the example previously cited.

Increase system use to two 3 credit courses each week or 6 hours of system use and the cost per student contact hour decreases significantly. The cost breakdown when offering two courses on the audio system would be:

AUDIO

Audio to four campuses
-3 credit course to 80 students
-another 3 credit course to 80 students

Student Contact Hours (SCH)
-80 students x 3 Hrs/Wk x 15 Wks = 3600 SCH
-80 students x 3 Hrs/Wk x 15 Wks = 3600 SCH

Cost
-System cost of $1,250/Mo
-$312.50/Wk ÷ 6 Hrs/Wk = $52.08/Hr
-90 Hrs programming x $52.08 = $4,687.49
-Salary for two 3 credit courses = $6,233.32
-$6,232.32 + $4,687.49 = $10,920.81

Cost Per Student Contact Hour
-$10,920.81 ÷ 7200 = $1.51/SCH

RESEARCH SHOWS SUCCESS

Evaluation of telephone-based instruction has been carried out with varying levels of scientific methodology. Some of the studies have compared telephone use with that of other media; others have not. Some were interviews with participants, other involved extensive statistical evaluations.

Following is a brief review of research projects in teleconferencing. It should not be considered a comparative or exhaustive analysis. Overall the studies indicate that telephone-based instruction is as effective as face-to-face instruction.

Nunley[1] (1965) studied the effects of telelecture in retraining two groups of elementary school mathe-

matics teachers. Pre and post tests of achievement, mathematical ability and attitude inventory were given to both the telelecture group and those in the conventional face-to-face situation. There was a significantly greater mean change in content mastery for the telelecture group than for the conventionally taught group. This difference, however, could not be accounted for by variances in ability, teaching attitudes or personal background.

Edelman[2] (1967) studied adult classes in reading and writing Hebrew. Two Hebrew classes were taught simultaneously to telelecture groups in Grand Rapids and East Lansing, Michigan, and to a live control group in Chicago. Each of the experimental groups was originally made up of a minimum of 20 students. A similar control group originally had 40 students. An analysis of scores on a midterm and on an oral reading test indicated no significant differences between performances of the three groups. The presence or absence of the instructor from the classroom did not affect the performance on any of the criteria.

Blackwood and Trent[3] (1968) evaluated 71 students enrolled in a short course on money management. All were members of county Extension Homemakers units in Reno County, Kansas. Pre and post tests were given simultaneously to both groups--the 34 who were taught in the traditional face-to-face method, and the 37 students who were instructed by telephone. Results showed no significant differences in the amount of learning which took place when the educational information was presented to the adult audience through face-to-face or remote teaching techniques and the personal and situational factors; age, level of education, time of day, attitude.

Boswell, et al.,[4] (1968) studied distance teaching involving three classes of an introductory psychology course at the University of Missouri-St. Louis. The 64 member control group received an on-campus lecture. A second group of 63, also on campus, received a live lecture which was simultaneously transmitted to the third group of 22 in a remote classroom. Pre test results indicated no significant differences in content knowledge among the groups and the post test also showed no differences among the groups after training.

Puzzuoli[5] (1970) studied the applicability of teaching university extension classes by telelecture and electrowriter. At West Virginia University, Morgantown, courses in mining, engineering and modern mathematics were given to college classes (control group) and also, using telelecture and electrowriter, to extension classes in a remote location (experimental group). Comparative analysis of the achievement of the two groups showed that achievement by extension classes was equal to, or significantly greater than, that of students in the on-campus classes. Analysis of extension student questionnaire responses supported the thesis that success in telelecture teaching is greater when the professor limits continuous lecturing to 20-25 minutes, provides printed material, uses audiovisual techniques as a supplement, makes a number of personal visits (two or three) to the class and bases the class on problem solving techniques.

From the research connected with the use of the telephone, other studies of interest on many types of subject matter include: graduate courses, Hartje[6]; undergraduate credit, Hoyt and Frye[7], and Zimmerman[8]; teacher education extension courses, Pellett[9]; formalized adult instruction at their place of business with the instructor rotating his appearances among the location, Gold[10]; and art education to small schools in four western states receiving simultaneous instruction from a central source with the aid of overhead transparencies, Jesser and Clarke.[11]

In summary, the studies indicated that the telephone-based instruction compares favorably with face-to-face classes. Consequently, the telephone is a useful educational communications tool capable of being used for effective educational experiences.

TELECONFERENCING HUMANIZING AND TEACHING STRATEGIES

In the century since Alexander Graham Bell put together the first telephone in Branford Ontario Canada, his invention has undergone vast technical development and is now connected to the world's most reliable and sophisticated nationwide network. Although the telephone has been an effective personal communications device, it has been long overlooked as an effective medium for education. The use of the telephone for education has been limited during the past 20 years in favor of more glamourous media. Thus it remains a virtually untapped natural resource for education.

The major thrust here will be to discuss some of the effective humanizing and teaching techniques that are built into the design of telephone-based instruction programs that can be integrated into many teleconference programs.

Program and Administrative Considerations

Let's examine the three most common limitations of teleconferencing.[12]

1. Making A Listening Group A "Group"

When 165 people are dispersed throughout a state at some 10-20 different locations, for example, it's difficult to make the listening group a "group" and not just a collection of strangers. However, the instructor can create a cohesive group feeling and group rapport by:

a. Always calling participants by name and responding to their questions or comments by name.

b. Travelling to teleconference locations to originate programs.

c. Sending out a half sheet of paper on which participants are invited to share information with or ask additional questions of the instructor if program time happens to run out.

These are simply techniques to let participants all around the state know that no matter how many miles separate them, two-way communications channels are always open.

2. Discussion Doesn't Just Happen

Adults like to get actively involved in the learning process. That desire for two-way communication, between participants and instructors and among participants, is the basic benefit of telephone-based

instruction as a viable, cost-effective medium. However, no matter how many adults want that opportunity and no matter how easy it is technically, some instructors may have to work at getting teleconference discussions started. There are a few basic reasons why participants may be reluctant to interact at first; they may feel they're infringing on precious program time, other people's time or their question or comment may not be worded correctly. On the other hand, adults frequently are more inclined to ask questions on a telephone system than in a normal classroom situation.

This interactive discussion can easily be initiated by the instructor by:

a. Asking participants to fill out an identification card so the instructor can use this information to help lead adults into discussion and to draw upon their personal experiences.

b. Asking a teleconference group at one site to work together preparing an answer to a specific question for the next week's session thus building in discussion for the next program.

c. Preparing participants for the teleconference learning experience so they're comfortable with it. That preparation may be in the form of a welcome letter before the series begins, explaining the teleconference equipment, how they'll use it and also how the course and its materials will benefit them.

3. People Don't Listen Very Long

It's hard work concentrating on a solitary audio message for very long. For adults, that "very long" is not much more than 12 or 15 minutes.

Because our brains can process information at a greater rate of speed than most people talk, pure sound requires a greater concentration from the learner. If the audio material isn't very challenging to the listener; if it doesn't change pace with a new topic, or a change of voice or an anecdote every 15 minutes or less, then it probably won't hold his or her interest. Minds will start wandering, looking for something more interesting and more challenging. Message retention drops.

There are a variety of techniques to improve remote interactive programming interest in order to stop student mind-drift. Teleconference instructors, for example, have found telephone-based instruction offers them considerable programming flexibility if they consciously address the liveliness of their message. For example, almost anything that catches and holds the positive attention of participants is worth doing. Although some of these techniques have been incorporated in adult education programs for years, there are unique adaptations which have proven especially successful with telephone-based remote education:

a. Substituting a panel discussion with varied voices and ideas in place of a single lecturer.

b. Guest lecturers brought into sparsely populated and remote communities via teleconferencing enable the students to exchange ideas, and ask questions in an across-the-table fashion.

c. Using case studies at each teleconference location for on-site discussion and analysis, followed by enegetic dissection by the entire class.

In summary, the limitations surrounding a non-visual, interactive, remote learning environment can be overcome. There are scores of techniques to humanize the teleconference learning experience, to stimulate statewide discussion and to make the programs lively and interesting.[13]

The questions of "How do I personalize this kind of learning experience for my students?" and "How do I get discussion going?" are among the first concerns expressed by new teleconference programmers and instructors. Although some of these techniques will be described, instructors are only limited by their individual imagination and creativity.

Learning in a remote environment may, at first, seem impersonal to some, but instructors can use several techniques to make the experience highly personal.

Personalizing the Experience for Students

Send Participants A Welcome Letter

Once a participant has registered, instructors should send a "welcome aboard" letter before the first class session. In the letter, the instructor would explain the teleconference system and course objectives. A first assignment can also be made in this letter to get participants involved right away.

Include a Picture of the Instructor/Guest Instructors

Participants are curious about their instructors and each other. Many lecturers have included a picture of themselves and brief bibliography either in the course brochure or in the handout materials for the first program. Occasionally teleconference programs that involve selected sites have asked participants to send a slide of themselves or the group at their site so it can be duplicated and included as part of a program session. In a program devoted to individual or group reporting, the combination of a person's voice with his or her picture can be an effective personalizing experience for all learners.

Distribute a Class Roster to All Participants

Just as participants like to know something about their instructors, they also want to know about each other. One way this can be accomplished is by use of a class roster. On the roster, participants can be separated by location, and included with their names could be a brief description of their educational and/or professional experiences, address and telephone number.

Call on People by Name

Most people like to be recognized by their name. When participants ask questions or make comments, they frequently do so by first identifying themselves and their location. This helps both the instructor and teleconference participants identify a name with a voice. Referring to someone's professional experience, for example, is another way to personalize the

teleconference learning environment.

Assign Reports

Participant reporting has many values, as evidenced by its inclusion in the format technique section and the discussion techniques. It is also an excellent way to personalize the teleconference experience for learners as they feel confident drawing upon their own experiences.

Establish Additional Communications Channels

One University of Wisconsin instructor regularly used an "Q and A-O-Gram" technique. This half sheet of paper was an invitation to students to make comments or ask questions that may not have fit into a regular program's question and answer period. Other instructors continually urge participants to write or call them if they have questions after a teleconference.

Visit Teleconference Sites

Many instructors try to visit as many different teleconference locations as they can during a program series. At this time, instructors should try to stay for an hour or two after the teleconference to further get to know their audience.

Schedule Face-to-Face Group Conferences

Instructors can also arrange one day face-to-face conferences in the middle or at the conclusion of their teleconference program series. For ease in student traveling, these conferences can be split among geographical areas, for example, or held in conjunction with other professional conferences. This conference allows participants to meet each other and to take part in a variety of learning activities that may not be able to be accomplished during teleconference sessions.

Let Personality Come Through

Finally in a telephone-based, remote learning environment, it's important for instructors and participants to let their personalities come through the system. Participants can and should hear the emotional attitude in each other's voice and know, for example, whether or not a person is enthusiastic and sincere by the way they come across on the speakers. When people smile, for example, their voice reflects it. If instructors are sitting alone in an office or studio, for example, they may feel their gestures and expressions are ridiculuous, but they're definitely not. Any animation that makes the message more vital is actually more important in a teleconference program than in the classroom.

Stimulating Statewide Discussion

Stimulating interaction on a telephone-based educational delivery system can be a challenge for instructors. Instead of instant discussion during a question and answer period, instructors may sometimes find themselves greeted with dead silence. To insure teleconference programs are truly interactive, instructors must seek opportunities which encourage participants to actively participate rather than listen passively. Here are a few of the techniques used:

Plant Questions

Planting questions among participants is one technique to get discussion started. Instructors have primed a few enrollees in their programs before a series begins to enter in with questions if discussion lags or is slow to start. Also, moderators try to have several questions handy themselves and feel comfortable asking them.

Assign Feedback

Programmers can create situations to generate questions and discussion. Frequently this is done with handouts that accompany the program. An instructor, for example, may ask two or three teleconference locations to be ready to react during the next program to several specified questions.

Relate Questions to Participant's Occupation

By having the names and occupations of participants in front of them during a program, instructors can design discussion around this information. These facts can easily be obtained from the course registration blank or from participant identification cards completed at the first teleconference program.

If an instructor happens to be talking about apprenticeship, for example, and he knows from his list that there are five to six people in his teleconference class who either work with an apprentice or who are associated with an apprenticeable occupation, then he can direct his question to some of those people. In addition to stimulating discussion, this technique also helps personalize the learning experience.

Initiate Group Projects/Weekly Quizzes

Group projects or weekly quizzes completed by each site can be used to stimulate both on-site and network discussion. Both will permit learners to actively participate in a face-to-face manner and will also keep the program content relevant and practical. Some instructors find a weekly quiz at the beginning of class is an excellent stimulant for discussion on a teleconferencing system.

Group to Group Discussions

Not all questions and comments need to be channeled through the teleconference instructor or moderator. Participants at one location, for example, can talk directly to or question students at one or more other distant locations. Group projects are one way to create situations where participants share discussion among themselves. After hearing or reading a group project report from one location, another group could be assigned the task of evaluating the report on the network or further questioning them. Other sites around the state may join in this discussion and the moderator/instructor can step back and join in the discussion as a fellow participant.

Schedule Oral Assignments

Short reports of five to ten minutes can easily be incorporated in program formats. In addition to stimulating interest among participants, this technique is helpful for triggering comments or reactions.

Teaching Techniques

The two most important elements of a successful program are: (1) content and (2) method of presentation.

Although instructors have incorporated a variety of teaching techniques into their programs for years, these approaches must be adapted to fit the needs of telephone teaching.

These formats include:

Lecture/Question and Answer

The lecture and question and answer format is a popular teleconference format. Doctors, nurses and attorneys, for example, are accustomed to learning this way, and they're a clientele group who like programs in this format. It's also an efficient method. Large numbers of facts, new information or review material can be presented in a short period of time.

Via teleconference systems, however, it's not just a lecture. The lecture is reinforced by supplementing oral material with written. Prior to a program, participants are sent skeleton outlines on which they can easily follow and take notes during the lecture. Significant visual materials should also be incorporated into handouts. A list of study questions, bibliographies, additional reading sources, and perhaps brochures, mimeographs, overheads or slides are among other forms of visual material that may be used by teleconference instructors to complement and reinforce their audio presentations.

Sufficient question and answer time is required to insure two-way communication. Teleconference participants need the opportunity to clarify speaker ideas, support or dispute material, or carry the discussion into personally relevant areas.

Celebrity Expertise

Well-known local, national or international authorities are available as teleconference guests. Guests of this caliber not only attract participants to the program, but they also are rich teaching and learning resources and, of course, interest-holders. Guests can easily and inexpensively be connected to teleconference networks via long distance telephone lines. Main presentations may be given live during the actual program, or prerecorded. In the later case, guests can be connected live for the question and answer period. All speakers should be provided with an audience profile. Speakers should also include skeleton outlines or other handout material to reinforce their audio presentation.

Participant Reports

The peer teaching concept is a teaching technique effectively incorporated into teleconference programming. Here students share their experiences as they relate to the subject being discussed with other teleconference students and consequently assume part of the teaching responsibility.

Interview

Interviews of 15 to 20 minutes in length are meaningful segments of teleconference programs. An instructor can either choose a resource person to interview or ask for class participation in the selection process. During the program, the instructor interviews the guest with five to eight questions that participants have sent him. Following that segment, discussion is opened to all remote locations.

Interview formats can also be successful if a particular guest doesn't have time to tailor his or her ideas to meet the needs of a specific audience, or if the guest is not fully aware of audience background or previous teleconference program content. The course moderator/instructor in this case could act as an interviewer, asking the speaker questions to help him or her respond in a relevant manner.

Panel Discussion

If the topic is complex or controversial, panel discussion can create lively, informative teleconference programs. Panelists may be located anywhere in the state or the country. In addition to the topic at hand, the variety of voices helps hold listener attention. Panels should include not more than four members, however, as participants need to be able to distinguish panelists' voices. And, as with any panel discussion, a moderator is essential for keeping the topic in focus. Again, bibliographies and any other appropriate written materials should be sent to the students prior to the program.

Reactor Panels

Program communication, particularly clarification of important points and ideas, can be improved through the use of reactor panels. Members of the panel would be the initial reactors to a teleconference speaker's presentation, guest lecturer, interview or panel discussion, representing the entire audience. To be successful, the speaker, reactor panel members and moderator should agree on program objectives and procedures before the actual program.

When large teleconference enrollments and limited program question and answer time are a problem, reactor panels can be an efficient means of covering common questions that may be on many participant's minds. Once these basic questions are covered by the reactor panel, the network could be opened up for questioning by all participants. This technique is also successful to stimulate discussion.

Role Playing, Skits

Small group skits or playlets can stimulate audience thinking and teleconference discussion. For example, one program for social workers and foster parents frequently uses short vignettes, for example, to realistically portray foster parents feelings. Dramatic talent isn't required. Participants can prerecord skits prior to the actual program or peform them live during the program. On teleconferences, it's best to stick to structured role playing, where the

script and performance are carefully planned out prior to the presentation. A suitable introduction and discussion following the skit are needed to insure the intended objective was met.

Buzz Groups

Teleconference participants frequently split into location buzz groups during programs. For example water supply operators from numerous communities used approximately 20 to 30 minutes of their two hour programs for local discussions. After the program's main presentation, groups "broke off" from the network to discuss how their location situation related to the day's topic. This buzz group format gave participants an opportunity to actively participate in the program and also helped keep the program content relevant and practical. A group leader at each location served as a resource person to answer questions, thus alleviating the need to use limited program time for some participant questions. Buzz sessions can also be used to formulate group questions, brainstorm a problem or develop a case study presentation or solution.

Case Study

Students learn by working on real problems. Case study examples are one way to bring discussions down to earth and to stimulate both on-site and network discussion. For example, a group of clergymen enrolled in the program Parish and Personal Conflict: Disaster or Opportunity were encouraged to bring in written case studies of conflict situations in which they were currently or had recently been involved. This data then became the agenda for site group consideration. Spokesmen at each site represented their group and reported on how their group analyzed the problem.

Group Work Sessions

Many programmers structure their sessions to include times of activity as well as listening and discussion. An agricultural engineer who frequently works with 4-H woodworking leaders via teleconference does just that when he presents technical kinds of materials. In a wood identification program, the first half of the program laid the basic background information to the topic; the second half was devoted to actual wood identification by participants at each site.

A wood identification key and many samples of wood had been sent to each site. The program instructor would work with the teleconference group by saying, "Okay, pick up the piece of wood marked number one. Now let's look at it and tie it into the key. Does it have this characteristic, does it have that characteristic?" After working down the key for practice, he'd ask participants to take a second sample and trace it down the key. When participants at a site felt they had it identified, then they'd come on the network and try to get some agreement from others at other sites.

Brainstorming

Brainstorming is a simple concept for getting the most creative energy out of a group discussion. Teleconference committee meetings often follow this format. Here the purpose is to collect a large number of ideas, and brainstorming sessions on the network have ranged from collecting ideas about heritage programs to ideas for a weekend conference for forest owners.

After the program rules are explained, moderators simply open up the network for ideas. The quantity of ideas is more important here than the quality. Freewheeling should be invited. Editing and synthesis can always be accomplished after the session. Follow-up teleconference sessions can also be scheduled to appraise and further evaluate edited ideas.

Pre-Teleconference Warmup Sessions/
Post-Teleconference Sessions

One method used successfully by several teleconference clientele groups is a warmup session prior to the actual program. By assembling groups at sites an hour before the teleconference, participants can work together on group projects, case studies, discuss outside reading material, and so on in small face-to-face situations. Often these groups find this presession to be valuable warmup for getting into the topic. Then, when it's time for the program to begin, they are eager to share their ideas, thoughts and comments with other participants and the instructor.

On-site discussion can also be scheduled for the hour following the session. Participants at each location use this time for such things as group projects, brainstorming, etc.

Summary

Teleconferencing is only a means to an end. Initial psychological barriers to teaching via a telephone-based educational delivery system will probably be more difficult for instructors to overcome than the real ones. However, once these limitations are understood and dealt with, the many capabilities and potentials of telephone-based remote learning will become evident.[14]

Evaluation Process for the University of Wisconsin's Educational Telephone Network (ETN) Programs

In order to determine if the appropriate educational techniques were designed into programs transmitted via Wisconsin's ETN, an evaluation system called the Program Evaluation Checklist (PEC) was developed. Its purpose is to study the relationship between the telephone system of communications and the educational objectives of a program. Specifically, it is concerned with tailoring educational materials for delivery to multiple remote listening stations via a telephone-based system.

The development of this evaluation system required a cooperative effort between content specialists and communication education specialists. This joint effort lead to an agreement concerning a number of variables common to the concerns, interests and objectives of both disciplines.

The PEC consists of a list of statements which defines specific aspects of program content, organization, methods of presentation, materials, lecturer and technical aspects. The participants, representative of an entire ETN class, are asked to evaluate each state-

ment and determine how well it describes the program. For example, in evaluating the program lecturer, one question asks, "The program lecturer/moderator encouraged participants to take an active part in the program." Ratings are made on a numerical scale of 0 to 9. Demographic questions are also included in evaluations to obtain profiles of ETN students.

At the completion of the evaluation process for selected ETN programs, an extensive report is completed and is reviewed with individual instructors. Evaluation materials are strictly confidential between the Instructional Communications Systems staff and the instructor. The paramount purpose of this evaluation process is to determine program strengths and weaknesses and to assist and enhance the selection and utilization of appropriate educational techniques for future ETN programs.[15]

THE TECHNICAL COMPONENTS OF TELECONFERENCING SYSTEMS

A successful teleconferencing system whether made up of two or 200 locations includes three basic components. These are: (1) the terminal or station equipment, (2) the interconnecting transmission system, (3) the network control center. Many variables influence the design of a teleconferencing system. Included are the number of locations; the geographic area covered by the interconnect system; the network configuration, whether the system is a four-wire multi-point dedicated network or a two-wire plain old telephone service (POTS) dial-up network, and the network's intended use, whether designed for corporate committee meetings or statewide outreach education. No two systems will likely be duplicate in design, but several ingredients and operational capabilities are common to successful teleconferencing systems.

Common Technical Ingredients

A teleconferencing system should be able to connect participants at widely scattered regional, statewide or national locations.

The system should provide a communication environment that duplicates [as closely as possible] the single site discussion group.

Terminal or station equipment at locations should provide even sound distribution for up to 200 people and allow easy participation.

The system should be manageable, allowing location and equipment additions and changes to meet user's changing needs.

A network or system control center should be maintained to supervise the network's technical operation and provide a program origination point.

The transmission system should provide clear, intelligible communication between all points on the network.

The system should be able to connect groups anywhere in the world via the regular telephone service.

Methods should be developed to quickly recognize and clear any technical problems interferring with service.

Designs and associated equipment should feature the latest technology and be reliable and serviceable.

All terminal equipment, station location interfaces and installation procedures should be standardized throughout the system.

With these general operational capabilities in mind the three basic components of a teleconferencing system will be described in greater detail.

Terminal or Station Equipment

The following criteria are requirements for successful teleconferencing terminal equipment:

1. The equipment must have four microphones to cover up to 12 people at each location. Microphones must have sufficient cord length for easy reach by participants. Approximately 20 feet of microphone cord can meet varying room and equipment placement conditions.

2. Up-to-date technology must be used in microphone selection. A high quality cardioid microphone will minimize room, acoustical feedback and noise pickup.

3. The speakers must exhibit a high degree of intelligibility to cover up to 200 people at each location.

4. The equipment must produce uniform volume throughout locations, independent of room acoustics. Additional plug-in column speakers should be available for very large or irregularly shaped rooms.

5. A push-to-activate switch should be on each microphone. This switch should be spring loaded so that the microphone is deactivated when the switch is released. This switch does not control the microphone in the classic push-to-talk sense. Experience has shown that participants have difficulty coordinating their hand in a push-to-talk and release-to-listen function. Therefore, voice operated attenuators should be provided in both the receive and transmit signal paths in the station equipment. These voice operated attenuators serve two functions; first, they eliminate potential feedback loops between stations and, second, they allow participants to communicate over the system without the restriction of the classic push-to-talk function.

When a participant depresses the push-to-activate bar, this activates his particular microphone but does not cut off the speaker. At this point the system is in a voice activation mode. The voice activation mode allows fast switching of receive and transmit paths and provides maximum interaction over the system.

In contrast, when there are more than 10 locations on a system the presence of 40 microphones all "open" at once causes noise pickup from chair rattling, coughing, paper shuffling and so forth that makes the network unusable. Unfortunately, unlike the human ear, the microphone cannot distinguish between voice and other room noises. In such a system, normal room activity causes the microphone switch to trip. The speaker will cut in and out as the switch operates and information will be lost.

A combination of a push-to-activate switch on each microphone and the voice activated switching mode is a successful compromise for interaction capability

COMPONENTS OF A TELECONFERENCING SYSTEM:

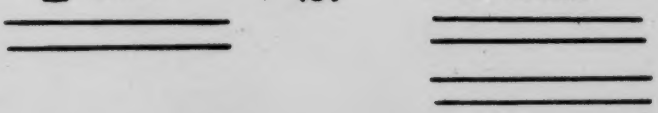
- **STATION EQUIPMENT**

HANDSET (2 WIRE)

VS. "LOUDSPEAKER TELEPHONES" (2 OR 4 WIRE)

- **TRANSMISSION SYSTEM**

2 WIRE VS. 4 WIRE

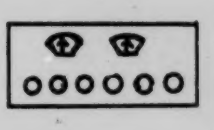


- **CONTROL CENTER**

"OPERATOR-ASSISTED" CONFERENCE CALL

"DIAL-UP" CONFERENCE BRIDGE

"MEET ME" CONFERENCE BRIDGE



TECHNICIAN-CONTROLLED CONSOLE

on a teleconferencing system. The push-to-activate switch eliminates the accumulated room noise problem and the voice activated directional switch eliminates participants dealing with the classic push-to-talk function.

6. Side tone should be built in the equipment to assist in proper microphone operation. A local public address feature is an example. Its level is set in relation to the incoming program level so participants receive proper side tone level when speaking in a normal voice four to six inches from the microphone. Side tone can be used as a public address feature when the equipment is not used for network programs.

7. The equipment should be designed with standardized interface elements that are uniform with telephone company regulations. The equipment should use the latest technology to provide long term reliability.

8. The terminal equipment should include a local test feature so participants can verify that the equipment is operating properly. A loop back test feature performs this function. Combined with a total system troubleshooting plan, this feature streamlines fault detection and repair. For example, if a location with trouble uses the loop back test feature to verify that the station equipment is operating properly, it eliminates a repairman traveling to the site to check the equipment. The result in time and dollar savings is significant.

9. The terminal equipment should be able to add microphones. An outboarded mixer with microphones that plug into the standard station equipment provides this feature. Mixers can provide additional microphones for locations with large numbers of participants.

The University of Wisconsin uses station equipment designed and manufactured by Darome, Inc., Harvard, Illinois, for its 200 point dedicated network. The equipment has been highly accepted by ETN students and has proved extremely reliable for over three years.

Teleconferencing Transmission Systems

Two general types of teleconferencing transmission systems are in use today. These are a POTS two-wire dial-up network similar to an operator assisted conference call, and a multi-point four-wire dedicated network of leased telephone company facilities. In general a dedicated multi-point conferencing system will utilize the POTS two-wire capability to extend its coverage points to resource people anywhere there is a telephone.

Plain Old Telephone Service (P.O.T.S.) Two-Wire Conferencing System

The two-wire teleconferencing system uses the available public telephone network as a transmission system for its interconnects. Each selected station is dialed individually and connected through a bridging arrangement before the program. Many such conferences are regularly conducted with 10 or less locations using available telephone company conferencing facilities.

A two-wire system must meet three critical requirements to provide successful teleconferences. These are uniform terminal or station equipment, effective bridging for even sound distribution and uniform signal levels at all locations, and efficient controlling functions that allow rapid set-up, location selection and troubleshooting.

A successful two-wire bridging arrangement must also have the capability to connect up to 30 two-wire telephone lines together. The use of up to 30 point two-wire dial-up conferences is a resource just now being developed. Research and development efforts are being conducted in both Bell Telephone Laboratories and in the private sector to provide an efficient and effective multi-point two-wire conferencing capability.

Multi-point Four-Wire Dedicated Teleconferencing System

A multi-point private line network utilizes existing telephone facilities to interconnect its many locations. A four-wire system of dedicated lines allows stricter control over transmission system parameters that affect the quality of signals on the system. Fine tuning techniques on such a system results in a transmission quality far exceeding that from a two-wire conference system. The main reasons for this improved transmission quality are stricter control over signal band pass, signal levels and signal to noise ratio. Transmission systems engineers for the Wisconsin Telephone Company have developed criteria for establishing a successful private line multi-point teleconferencing system. For greater detail refer to Braun, Gilbertson and Hansen, 1976.[16]

The basic building block of a four-wire multi-point private line network is a four-wire, six-way bridge. Using this in combination with existing telephone transmission facilities a network of any size can be designed. These bridges can be located regionally; thus a network can be formed in a regional building block fashion. The practical size limit for any region of a multi-point private line telephone network is 20 stations because of noise created by the multiple facilities and circuit terminations.

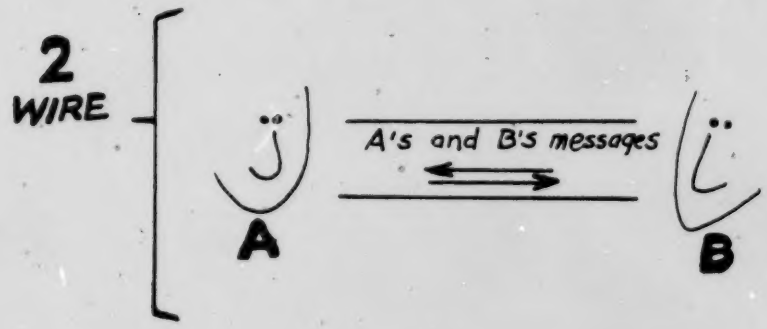
The University of Wisconsin's 200 station teleconference network operates with satisfactory noise levels because it is a combination of 10 noise-isolated regions. Each region serves approximately 20 stations. The regions are isolated by using voice gated amplifiers at the master combining bridge, located at the central control point. This regional arrangement limits noise experienced at any station to that generated by facilities and terminations within the region.

Most of the techniques used to construct a multi-point private line four-wire teleconference system are standard operating procedures within the telephone industry. Equipment is available from several manufacturers and suppliers of standard telephone industry equipment.

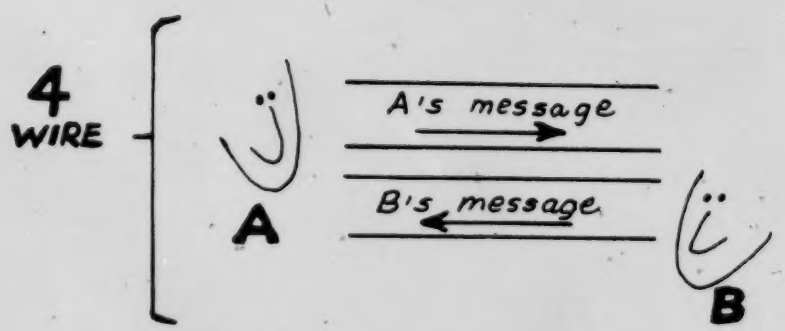
An important ingredient in a successful private line telephone network is the ability to connect stations on the public telephone network to the four-wire system. This involves interfacing a two-wire dial-up teleconferencing system with the usually large private line network. The University of Wisconsin system can connect up to six POTS lines to its 200 point network. These lines connect resources who cannot be at network location to the statewide audience.

TELECONFERENCING STATION EQUIPMENT

2 WIRE VS. 4 WIRE TRANSMISSION



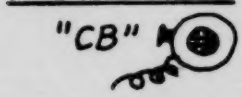
- A "BASIC" CIRCUIT CONSISTS OF A PAIR OF WIRES.
- MESSAGES ARE "MIXED" — CHANCE FOR FEEDBACK ON "RETURN LOOP."

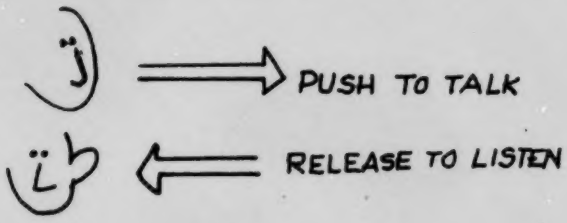


- EACH MESSAGE CARRIED ON ITS OWN PAIR OF WIRES.
- MESSAGES ARE "ISOLATED" FROM EACH OTHER — NO CHANCE FOR FEEDBACK.

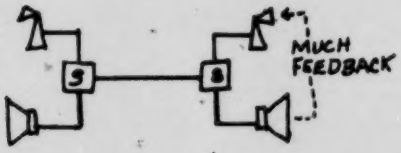
TYPE OF SWITCH – CHARACTERISTICS

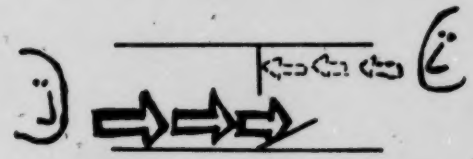
MECHANICAL "CB"





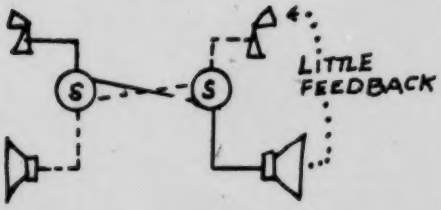
VOICE-ACTIVATED HARD SWITCH "2 WIRE"

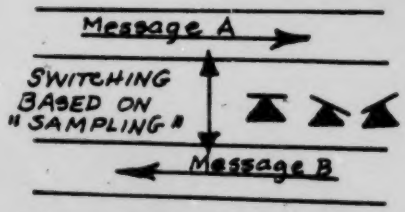




LOUDEST VOICE GETS THROUGH – "ALL-OR-NOTHING"

VOICE-ACTIVATED SOFT SWITCH "4 WIRE"





DEPENDING ON WHICH PAIR OF WIRES HAS LOUDEST MESSAGE, OTHER PAIR WILL BE "SOFTENED" BUT NOT COMPLETELY CUT OFF.

Some other keys to the successful development of the University of Wisconsin's private line system include:

1. Limiting amplifiers are used to equalize signal levels between the private line network and the public telephone network (POTS).

2. The network line up levels are optimized for best signal to noise ratio.

3. A master bridge combines the regional networks.

4. The regional control circuits utilize voice gated amplifiers to isolate region noise levels.

5. Standard bridging configurations and equipment minimize design and maintenance practices.

6. All station terminations are standardized, passive and hard-wired using a reliable four-prong jack.

7. Telephone company personnel receive uniform station installation instructions to insure standardized location installations.

8. Uniform testing instructions, one trouble reporting center, and special loop back test features in the portable station equipment facilitate speedy and effective trouble detection and repair.

The exact makeup of any conferencing transmission system depends on many variables, including the available telephone facilities within a particular region. The specific network design should not be constrained by available telephone industry hardware or other standard operating practices. The design and implementation of any system should be sensitive to user requirements, although these requirements are not always met easily. History has shown that highly successful teleconferencing endeavors occur where users and telephone industry personnel work as a team to provide the best service possible.

Teleconference Network Control Center

The control center provides technical supervision and control of the teleconferencing network. It also facilitates network programs.

Any teleconferencing effort more ambitious than an occasional four to five party operator assisted conference call requires an identified control center. This network control center may simply feature standard station equipment with a telephone to report troubles. It may include a tape recorder to record programs. The control center may also be as complex as the one used at the University of Wisconsin where outreach programming runs 16 hours a day, six days a week.

Regardless of the size of the telephone conference control center, four major functions are involved in complete operational control of the system.

Program Control

The control center operator uses a console to control audio sources such as tape recordings and microphones that are used as part of the program. For example, the operator loads any insert tapes on the tape machines and plays them on cue from the instructor. The operator also maintains control over program signals and records the program for future reference. During the program the instructor does not have to cope with the hardware. Using the control center equipment, all technical aspects are handled by the operator.

Telephone Network Control

The telephone conferencing network may be totally dial-up, dedicated, or a combination. The dial-up network uses regular two-wire telephone service connected via a special bridge arrangement. Prior to program time, an operator uses controls that may be similar to a call director and automatic dialer to quickly establish the teleconference network. The network of dedicated lines used in a multipoint private line service, can be arranged as programs and enrollments dictate. Participating locations can be selected for total system programming or regional networks can be established and programmed simultaneously. Dial-up bridging capability can extend coverage of a dedicated network to anywhere in the world via the public telephone service. Complete control of the entire network is maintained in the network control center. The particular control arrangement depends on the design and makeup of the transmission system.

Studio Sound System Control

This provides an important monitoring function for the network control center. Sound distribution is provided to three key areas: in the studios so the instructor can hear the response from the network, in the control room so the operator can monitor all audio channels, and to administrative offices so center staff can listen to programs. General building paging with emergency override to all areas is often included in a large comprehensive network control center.

Network Failure Detection

Effectively detecting and resolving system failures is a key to the success of any teleconferencing network. At the University of Wisconsin system failures occur as few as one or two times a day or up to as many as 10 to 15 times in a 16 hour programming day. Three to five system failures is the daily average. It is important, therefore, that a control center operator isolate and identify these problems quickly. Adequate controls and testing features are consequently integral to the control center. If the network control center operator can isolate problems in the equipment or the telephone services before calling in the trouble to the telephone company, the problem can be solved quicker and easier. Not only can this operator isolate the difficulty, but with experience he can identify the more specific problem, whether its station equipment, transmission line, carrier system, or some other characteristic problem. Given the ability to quickly isolate a region the operator can bypass the problem area and continue to operate the remaining parts of the network. Adequately equipped and manned with trained technical personnel, the control center can minimize network downtime and program interruption caused by system failures.

It is important to recognize that both the customer and the telephone company benefit by providing as many troubleshooting and problem solving controls and

features as possible in a network control center. These features help the control center operator minimize the effect of system failures on ongoing programs. A secondary benefit is that this wide range of trouble detection features allows the operator to do most of the troubleshooting--saving the telephone company time and repair costs.

Control center operations at the University of Wisconsin illustrate this process. This network is a good model because it has grown from a few stations to 200. Consider the 10-15 station teleconference networks currently being planned or implemented. As their usage and locations increase, the need for controlling the technical operations will also increase. Eventually, these networks may become as large and their control centers as complex as the one described here.

The University of Wisconsin control complex provides operational control of the 200 point network. Within the complex are studios from which instructors originate their programs and a control room which is manned by a trained technician who handles all equipment and technical adjustments during each program.

The instructor who originates programs at the control center has to cope with minimum technology. During his presentation he need only speak into a microphone controlled by a technician. He hears comments or questions from the network through studio loudspeakers.

The operator uses studio and control room equipment to guarantee a smooth trouble free program. Because he runs all technical aspects of the program, the instructor is free to concentrate on his presentation. The operator's role can be illustrated by describing the steps involved in setting up and operating a typical program.

Before a program, the operator refers to a daily schedule and program format book similar to a radio broadcast continuity sheet which outlines technical operational requirements. He arranges the studio with enough microphones and chairs for participants and sets up a slide projector and screen if needed. Insert tapes are loaded on tape machines and another tape machine is set to record the program.

When the instructor arrives the operator greets him and reviews the program format to check any changes. If a resource is to be connected to the network via telephone, the operator places the call, gets the party on the line, and when ready, connects him to the dedicated network. A program introduction on cartridge tape is played. Studio microphones are switched on and the instructor begins.

During the program the operator plays any tape inserts on cue from the instructor. If a problem develops with the control room equipment, he quickly transfers to a backup unit. If he hears trouble on the network, such as noise, he uses controls to isolate the offending region and reports it to the telephone company test board. All efforts are made to minimize the effect on the program. Thus the operator helps to isolate and correct the problem. Most of the time problems do not occur, however, and the operator continues to monitor the program. This is a simplified view of what occurs in a large teleconferencing network.

The scope and complexity of the control center is dependent upon network size, the clientele being served and programming demands. The large University of Wisconsin network operates as a service agency providing statewide teleconference capability to diversified clientele. The nature and content of the programs being offered by these groups is equally diversified. The program support requirements, system capabilities and control versatility necessary to meet the changing demands of programmers increases the features and complexity of the control center. Characteristically in large systems no one special interest group can support a network the size of the one at the University of Wisconsin. Many beginning networks are motivated to serve single interest groups but are forced to diversify and set up the system as a service agency to spread the operational costs over a large user group.

In the last several years the number of smaller networks has increased. Generally these networks tie two to twenty meeting or conference rooms together in companies, banks and federal agencies. The size of these networks is established initially and locations added usually on a temporary or dial-up basis. These small teleconference or corporate meeting networks may require only a minimal control center. This mini control center would be operated by a user or participant in the meeting rather than a trained technician. However, the same types of functions outlined earlier are required here.

To illustrate this let us describe how these functions relate to a minimal control center.

Program Control

May consist of a cassette machine that can be cued up to present prerecorded programs for inserts. Another cassette machine records the entire program. Copies of this tape can be made and distributed to the participants as meeting notes.

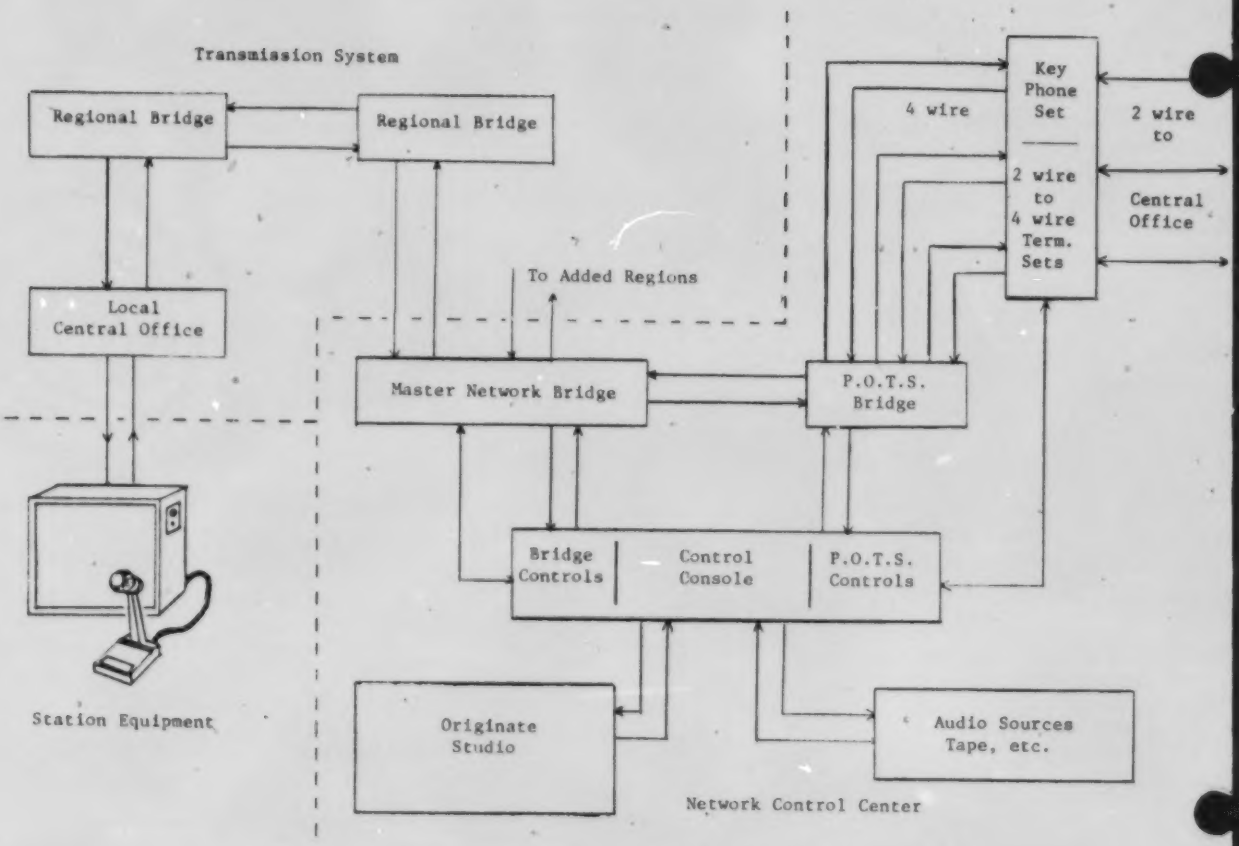
Telephone Network Control

Consists of a telephone with the telephone lines assigned for use on the teleconferencing bridge. The operator can be the user of the network and can simply place the call or calls required to complete the telephone network for the conference. This portion of the control center would resemble a call director. After the calls are completed the operator ties these lines into the conference bridge using appropriate controls on the call director panel.

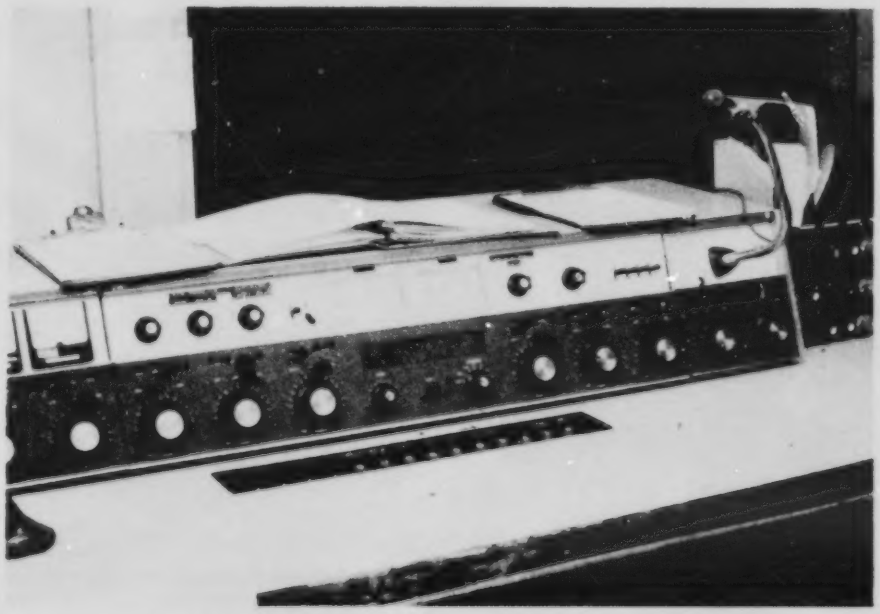
Network Failure Detection

The control center will have an additional telephone number reserved for trouble calls from participants. This trouble number allows participants to reach the control center operator to describe their problem when they may not be receiving the program or be able to transmit on the network. These simplified control center functions are displayed in a nontechnical manner so that the operator of the teleconferencing system can also be a participant.

There is a tendency for networks that have met with initial success to grow. Sometimes this growth is based strictly on the need for program material. Many technical and operational problems may be tolerated because of these needs. Some midwest networks are large, for example, but are almost unmanageable from a technical standpoint. These networks do not have a control center and are generally operated by the customers without any trained technical support. Consequently, when a problem occurs on these networks, the



Components of a Teleconferencing System



NETWORK CONTROL CONSOLE

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telephone company is contacted to isolate it. Generally, this disrupts programs and causes significant downtime of the network. It is costly for both the telephone company and for the customer who normally charges students for the course. Many of these network problems could be resolved with a control center that provides the features described for the Wisconsin system.

A teleconferencing network should start with a control center that will grow with it. Rapid growth and changing demands of network users require a high degree of versatility in the control center equipment. As more features are added, operational complexity increases. These functions are best handled by a trained electronics technician. So along with a well equipped control center, provision should be made for a technical staff qualified to operate and maintain it.

A control center equipped to meet the needs of a large network, such as the one at the University of Wisconsin, must provide more than functional control of the network. It also requires studios and a control room. Much of the equipment is there to augment or improve the instructor's presentations. Certain peripheral support equipment will be required, including tape duplicating and processing equipment, recording studio capability and repair shop support.

The heart of a teleconferencing network control center is the master control console. It is designed to provide a flexible and efficient means for performing the controlling functions of a network. The master control console switches and controls the output amplitude of various audio sources such as tape machines, turntables, studio microphones and telephone network feeds. The ability to mix, distribute, switch and control these audio sources is provided to assign each to its proper destination. Monitoring for program and audition is provided along with testing features and monitoring capabilities. Both four-wire and two-wire bridging modules with associated controls are included to connect regular telephone lines to the teleconferencing network. This master control console also provides audio production and recording capability to the network control center.

In summary, a teleconferencing network benefits greatly by having a control center. Some of these benefits are:

1. The instructor can concentrate on his presentation because he has a minimum of technology to cope with.

2. Maximum use can be made of available telephone circuits.

3. Troubles are recognized, isolated, and repaired quickly, resulting in less network downtime and minimum program interruption.

4. Technical aspects of the teleconference network, effectively controlled and supervised, contribute to the evaluation of a program's success.

ADVANTAGES OF TELECONFERENCING

Research on the use of the telephone in education indicates the telephone is a highly effective delivery technique in communicating educational materials to students throughout a large geographical area. Research also indicates there has been no significant difference between the performance of students with the presence or absence of an instructor from the classroom in teaching via a telephone-based system.

Having examined this and other aspects of telephone-based educational delivery, it seems fair to ask just what are its advantages for students, for instructors, educational planners, or for that matter, educational institutions:

For Students:

1. Teleconferencing provides truly continuing education enabling participants to receive limited amounts of information frequently in their home community. In this way, students can assimilate new information into their daily activities, unlike some of the more traditional forms of continuing education in which students receive massive doses of information once or twice a year.

2. Teleconferencing can meet the needs of students in remote locations. Some professionals have indicated they would not have settled in smaller communities if continuing education via a teleconferencing system had not been available.

3. Comparison studies indicate that students learn as much or more in teleconference situations as in other modes of learning.

4. Teleconferencing allows personalized communication between the student and the instructor and between students at a variety of locations.

5. Through teleconferencing, students can take advantage of quality education at reasonable costs, but are not required to travel great distances or spend large amounts of time away from family or occupational responsibilities.

For Instructors:

1. Teleconferencing allows a sharing of faculty resources. This faculty sharing allows schools to offer more complete educational programs and also allows outstanding teachers to reach students across broad geographical areas. For example, in a specific community college system, one professor taught 12 to 14 students in statics each semester. The other institutions in the system could not afford to offer statics as part of their curriculum. Through teleconferencing, one instructor now offers the course to all 14 campuses in the system. These classes would not normally have been available to students at other campuses.

2. There is great economy in both time and money for faculty. For example, they are not required to travel great distances or spend large amounts of time away from their primary teaching responsibilities.

3. Teleconferencing systems offer flexibility in program planning and scheduling. Systems can accommodate prerecorded tape materials, as well as live lecturers.

4. Teleconferencing systems provide multiple access points which allow instructors flexibility in program origination.

For Educational Administrators:

1. Teleconferencing systems enable institutions to reach entirely new audiences which have previously been neglected. For example, in one state a hospital program was offered to more than 400 housekeepers and 300 food service workers. Prior to this offering, there were few continuing education programs for these personnel. Even if there had been, it is doubtful whether the hospital would have released them from responsibilities and paid their expenses to travel to the courses.

2. Teleconferencing enables an institution to respond quickly and efficiently to the specialized educational needs of groups scattered throughout geographical areas.

3. Teleconferencing enables institutions to offer classes simultaneously to a limited number of remote locations. In other modes of instruction, this offer to few students at a location would not be feasible.

4. Teleconferencing allows institutions to share costs and faculty resources.

5. Teleconferencing systems may be used for administrative purposes when not used for instructional purposes.

6. Teleconferencing systems can be integrated into other types of instructional systems such as multimedia systems, computer assisted instruction and programmed instructional systems.

7. Teleconferencing is frequently less expensive per student contact hour, compared to other instructional media, e.g., remote face-to-face, video, CRI.

8. Teleconferencing has a primary advantage over most other instructional media in that it permits two-way instantaneous interaction between instructor and student and among students.

9. The public relations aspects of teleconferencing are significant. The programs presented via teleconferencing are tangible evidence that an institution is vitally concerned with the day-to-day problems of professionals and consumers, as opposed to the more esoteric research which may eventually filter down to the local level.

For Educational Institutions:

1. Teleconferencing can increase enrollments.

2. Teleconferencing can be responsive to the "needs of society."

3. Teleconferencing can bring quality education to rural areas with sparse population.

4. Teleconferencing can bring quality education to the underprivileged in core cities.

5. Teleconferencing can enhance the image of an institution through committee "branches."

For the most part, the use of the telephone for education has been virtually overlooked for the more glamourous media, e.g., video. The telephone remains virtually an untapped natural resource for education.

FOOTNOTES

[1]Bobby Gene Nunley, "A Study of the Effectiveness of Telelecture in the Retraining of Elementary Teachers in Mathematics" (unpublished Ph.D. dissertation, University of Texas, 1965) as recorded in Dissertation Abstracts, Vol. 26, N. 5, (Ann Arbor, Michigan: University Microfilms, Inc., 1965), p. 3165.

[2]Lily Edelman, "Teaching Adults Via Tele-Lecture and Electrowriter," Adult Leadership, October, 1968, pp. 163-64 and 189.

[3]Helen Blackwood and Curtis Trent, A Comparison of the Effectiveness of Face-to-Face and Remote Teaching in Communicating Educational Information to Adults, (Manhatten: Kansas State University Cooperative Extension Service, 1968). ERIC ED 028-324.

[4]John J. Boswell, Donald W. Mocker and William C. Hamlin, "Telelecture: An Experiment in Remote Teaching," Adult Leadership, March, 1968, pp. 321-38.

[5]David A. Puzzuoli, "A Study of Teaching University Extension Classes by Tele-lecture" (Morgantown: West Virginia University, 1970). ERIC ED 042-961.

[6]George N. Hartje, "Utilizing Telelecture in Teaching," International Journal of Instructional Media, Spring, 1974, pp. 219-26. ERIC ED 075-049.

[7]Donald P. Hoyt and David W. Frye, The Effectiveness of Telecommunication as an Educational Delivery System, (Manhatten: Kansas State University, School of Education, June 1972). ERIC ED 070-318.

[8]Harry P. Zimmerman, "Evaluation Inference Applied to Programming On An Educational Telephone Network--A Beginning" (paper presented at the Adult Education Research Conference, Minneapolis, Minnesota, February, 1970). ERIC ED 036-763.

[9]Vernon L. Pellett, "The Comparative Effectiveness of the Educational Telephone Network and Face-to-Face Lectures for University Extension In-Service Training" (unpublished Ph.D. dissertation, University of Wisconsin-Madison, 1970).

[10]Ben Gold, A Plan for Development of an Educational Telephone Network (ETN) to Extend Access to Educational Programs, (Los Angeles Community Colleges, Division of Educational Planning and Development, Office of Instructional Development, Los Angeles, California, October, 1973). ERIC ED 085-057

[11]David L. Jesser and Michael J. Clarke, Art By Telephone, (Carson City, Nevada: Western States Small Schools Project, 1966). ERIC ED 025-369.

[12]Lorne A. Parker, "Teleconferencing As An Educational Medium: A Ten Year Perspective from the University of Wisconsin Extension," in The Status of the Telephone in Education (2nd Annual International

Communications Conference), ed. by Lorne A. Parker and Betsy Riccomini (Madison: University of Wisconsin Extension, May, 1976), pp. 1-26.

[13]Marcia A. Baird, "Designing Teleconference Programs: Some Clues from the Wisconsin Experience," in The Status of the Telephone in Education (2nd Annual International Communications Conference), ed. by Lorne A. Parker and Betsy Riccomini (Madison: University of Wisconsin Extension, May, 1976), pp. 41-50.

[14]John S. Daniel and Ben Turpk, "Teaching by Telephone: A Two-Way Communication Mode in Distance Education" (paper presented at 10th International Conference, International Council on Correspondence Education, London, England, May, 1975).

[15]Harry Zimmerman, Hamilton McCubbin and Lorne A. Parker, "Program Evaluation Checklist (PEC)," University of Wisconsin Extension, Educational Communications, Madison, Wisconsin, 1968. (Mimeographed.)

[16]Daryl Braun, Dennis Gilbertson and Henry C. Hansen, "ETN: A Technical System," in The Status of the Telephone in Education (2nd Annual International Communications Conference), ed. by Lorne A. Parker and Betsy Riccomini (Madison: University of Wisconsin Extension, May, 1976), pp. 51-74.

ETN: A TECHNICAL SYSTEM

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ETN, Wisconsin's Educational Telephone Network, is a communication system designed to carry university extension programs throughout the state, and to do so with as little interference as possible with the on-going teaching and learning process.

ETN means a number of things to different people who use it. For the student in a remote area of Wisconsin, it is a means of access to college level and continuing education courses not otherwise available in the local community. To extension faculty members, it is a way not to travel. The instructor can deliver courses from his home campus to students throughout the state, eliminating the need to visit each community.

ETN is also a technical system. And to the engineers and technicians who designed and developed it, it is a network of telephone facilities, station equipment, and studio and control rooms that provide instantaneous voice contact between the student and the instructor. The specially designed station equipment is installed at approximately 200 locations throughout the state and interconnected by a network of telephone facilities. This interconnection uses approximately 5000 miles of voice channels and the service of 20 telephone companies to provide statewide coverage. Home base for the system is the Instructional Communications Systems studios and control rooms located in Old Radio Hall on the University of Wisconsin campus in Madison.

The development of ETN took ten years from its inception to reach its present state. Throughout the design and development of ETN, an important concept was applied to every facet of the system: The hardware must be as inconspicuous as possible, and every attempt should be made to minimize the technology that the students and instructors must cope with. Therefore, every effort was made during the design and construction of ETN to provide for easy operation and reliable trouble-free service.

The intent of this paper is to present a picture of ETN as a technical system. We will describe what exists today, users requirements, general design criteria and some of the problems and solutions encountered during ETN's development and implementation phases. The three major components of the ETN system, which will be examined here consist of the network origination studios and control room, the statewide telephone transmissions systems and the individual station location equipment.

ETN as a technical system hasn't always been as it exists today. Earlier attempts at teleconferencing in 1965 used a system much like a business conference call. A network was established through the telephone company operators using existing conferencing bridges. Station equipment consisted of standard desk top speaker phones. These early attempts were sufficiently successful to point out the need to provide programming on a statewide scale. This demanded a larger network with more stations tied together on a scheduled basis.

Transmission systems engineers for the telephone company pointed out the need for a major change in this approach to teleconferencing. On a large scale, the plain old telephone service (POTS), using operator assisted conference calls, was not a reliable means. Conferences were susceptible to many interruptions and delays in network hookups. The telephone company engineers proposed an alternative to POTS, one that would provide the dependable service required. They suggested a network of dedicated lines assigned exclusively to this service and available 24 hours a day.

The network was installed and new station equipment was provided. The new equipment consisted of a four wire telephone and handset with a 106B type speaker/amplifier. (Picture 1) This station equipment was installed in courthouses, hospitals, libraries and on the UW System campuses throughout the state. Locations were changed and installations added as programming and use dictated.

To use the new ETN equipment, a student would turn on the speaker and set the volume for a comfortable listening level at the start of each program. In order to ask a question or participate in the discussion, the student simply lifted the handset and spoke into it. Picking up the handset disconnected the loud speaker from the network. After having asked the question or made a comment, the student would push down the cradle button on the telephone or hang up the handset to allow everyone else at the listening location to hear the response from the instructor over the loud speaker. This equipment, tied together by the effective dedicated transmission system and controlled by the studio in Madison, served ETN users well and prompted rapid growth during the period between 1966-1968. Along with this growth came an increase in sophistication of the users. Students soon overcame the novelty of the new system, and instructors become more complex in their program planning. Users were no longer content with the mere fact that the system worked, and new demands were made of ETN. Technical shortcomings, previously overlooked because of the newness of the system, were no longer tolerated; they turned into glaring faults.

Analysis of system trouble reports identified a number of problem areas. The network's larger size and the resulting increase in transmission system circuit miles, caused the inherent line noise to rise beyond tolerable limits. Telephone system

engineers were able to control the noise build up by using the best available circuits and reconfiguring the network into smaller, more manageable blocks. Rapid and changing growth patterns made this a continuing effort.

Additions and innovations in studio and control room equipment kept pace with the changing needs of ETN as a program origination facility. The majority of trouble reports received pointed out shortcomings with individual station location equipment.

The 106B speaker/amplifier was small and unable to provide an even distribution of sound throughout many of the ETN classrooms. As class sizes grew it became difficult for people in large groups to hear the lecture. In order for someone to ask a question over the network, it was necessary for the student to pick up the handset and speak into it. The student would hear the response from the handset ear piece. However, picking up the handset from the cradle cut off the speaker, and in order for the rest of the people in the room to hear the response to the question, the student had to hang up the handset or depress the cradle button. This procedure was awkward and, as a result, made it very difficult for students to participate.

A plug and jack arrangement was developed and installed in various classrooms at ETN locations to provide versatility in room scheduling and arrangements. The handset and speaker were tied to additional equipment mounted at a central location in the building and connected with multipair cable. The multipair jack and plug required proved to be fragile. Resulting broken pins and plugs caused considerable downtime and an unusually high number of service visits by telephone repairmen.

A standardized equipment installation was needed to provide uniform operating instructions, trouble analysis and correction procedures. Though standards were established and detailed instructions sent with each service order, not all ETN station equipment was installed accordingly. One example occurred at a site with a very large classroom. During programs it seemed to take an unusually long time to receive any response from participants at this location. The reason proved to be because the speaker had been installed at one end of the room, and the handset on the wall at the other end. The students clustered around the speaker to hear the lecture, and, when they wished to ask a question, they dashed to the other end to use the handset.

The effort to standardize installations among 20 different telephone companies was an overwhelming one. No amount of instruction seemed to guarantee that each ETN location would be set up the same way. Long delays for new classroom installation were often encountered.

Attempts were made to alleviate some of these problems. Some solutions were only temporary, adopted in order to buy time to develop more permanent ones. A large speaker and amplifier was supplied by the telephone company and installed at locations with large rooms and class sizes. The package was called the "Webster teletalk" and is shown in Picture 2. This helped, but it was clearly not the answer. The need to make it easier to participate over the network still remained. The handset was too restrictive in a spontaneous interactive lecture situation.

About this time, the Bell Telephone Company introduced the 50A telelecture unit. 50A was designed for use on regular 2-wire telephone lines (POTS) and was employed by ICS to connect student groups from around the country to ETN via Direct Distance Dial (DDD). A modification to the 50A adapting it to 4-wire operation seemed like the answer. It provided the three main ingredients--microphones, a speaker, and a voice switch to relieve the user from having to push a button whenever he wished to speak. However, the 50A had some drawbacks. Notably, the speaker was small, which limited the size of the audience, and the set had just two microphones which could only be used one at a time. One microphone was selected to be made live by pushing a button on the set. This restricted its use in a roundtable discussion by slowing down the response from the group using the 50A. This equipment was a definite improvement over the telephone handset, but not the final answer.

Clearly, some major steps had to be made in the development of the technical capabilities of ETN. A Dean's Advisory Committee was formed to set up general operational requirements needed for a teleconferencing system such as ETN to successfully fill its role as a vehicle for Extension outreach programming. The result of this effort was the following general statement of operational capabilities:

ETN as a teleconferencing system should be able to connect student groups at a large number of geographically separated locations on a statewide basis. This system must provide a communication environment that duplicates, as closely as possible, a single-site discussion group. Equipment at each location needs to make it easy for students to participate and should provide even sound distribution for groups as large as 300 people. The system must be manageable, allowing additions and changes to be made as programming and use require. A central control and origination point should be maintained to supervise the network's technical operation and provide a point from which instructors can originate programs. The transmission system needs to be sufficiently noise free and of high enough audio quality to provide clear, intelligible communication between all points on the network. Ability to connect groups to ETN from anywhere in the world via the regular telephone service should be provided. Methods are needed to recognize and quickly clear any technical problems that may interfere with service. Designs and associated equipment should be of the latest technology, reliable and serviceable. All station equipment, station location interfaces and installation procedures should be standardized throughout the system.

With these guide lines established, an ETN project team composed of ICS personnel, Wisconsin Telephone Company system engineers and marketing representatives worked to implement them.

Existing ETN Station Equipment

Just as guidelines were developed for the entire ETN network, requirements for ETN Station Equipment were also outlined. In 1973, ICS personnel in conjunction with the Wisconsin Bell Telephone Company set forth guidelines which described a nonexistent station package. After interviewing several sound equipment manufacturers, Wisconsin Bell contracted Darome to repackage their existing sound system modules into the required station equipment.

In the fall of 1973, Darome, Inc. delivered prototype station equipment to ICS for evaluation on their 25 point SEEN Network during the balance of the 1973-74 university year. Experience led to refinements in operation and packaging which were incorporated into the Darome Model 604 which is the exclusive station equipment used on the SEEN and ETN networks operated by ICS.

Darome witnessed the educational impact of ETN and the need for similar conferencing equipment on POTS. The Darome Model 610 convener was designed to meet that need. ICS uses the Model 610 for interfacing instructors virtually anywhere in the world via the DDD network. These two models, the ETN station equipment, Darome Model 604, and the POTS station equipment, Darome Model 610, will be discussed here.

Darome Model 604

The Darome Model 604 convener is a self-contained, portable conference set and is intended for use on multi-point, private, 4-wire, voice grade telephone circuits. The 4-wire circuits have a transmit pair and receive pair to all locations. The unit consists of a station receive channel and a station transmit channel. The station receive channel, which processes the signal feed from the network that is fed into the classroom, consists of an input filter, preamplifier, 14 watt amplifier, and three speakers. This audio amplification system is sufficient to provide highly intelligible sound in a classroom seating up to 300 students. The incoming, or received, signal is adjusted by a single control which is readily accessible on the side of the unit. Care has been taken in the design of the station receive channel to minimize telephone line noise outside the speech bandwidth.

The station transmit channel, which enables students within the classroom to feed signals to the telephone network, consists of four highly directional desk top microphones, a microphone preamplifier, an electronic switch and a telephone line driver. To facilitate student participation, each microphone is equipped with a press-to-activate switch and twenty foot cords which allow the

MODEL 604 IN CLASSROOM USE

microphones to be placed around the classroom for easy access by students. With the microphone within reach, the student can respond to network programming by simply pressing the switch and speaking. A jack is provided on the basic unit to extend the number of microphones in a single classroom to eight with the addition of a Model 605 mixer. This process can be continued up to 12 or 16 microphones by using additional mixers. Microphone signals are not only fed to the network via the station transmit channel, but are also fed to the receive channel amplifier. Thus, the conference set may function as a public address system for its own room. The public address volume is set at a predetermined level in relation to the station receive signal. Single control operation of the unit minimizes set-up and acoustic feedback problems.

The Model 604 is housed in an attractive woodgrain vinyl cabinet which has clean lines for storage. Adequate space is provided in the cabinet for storing the microphones and their associated cables.

Darome Model 610

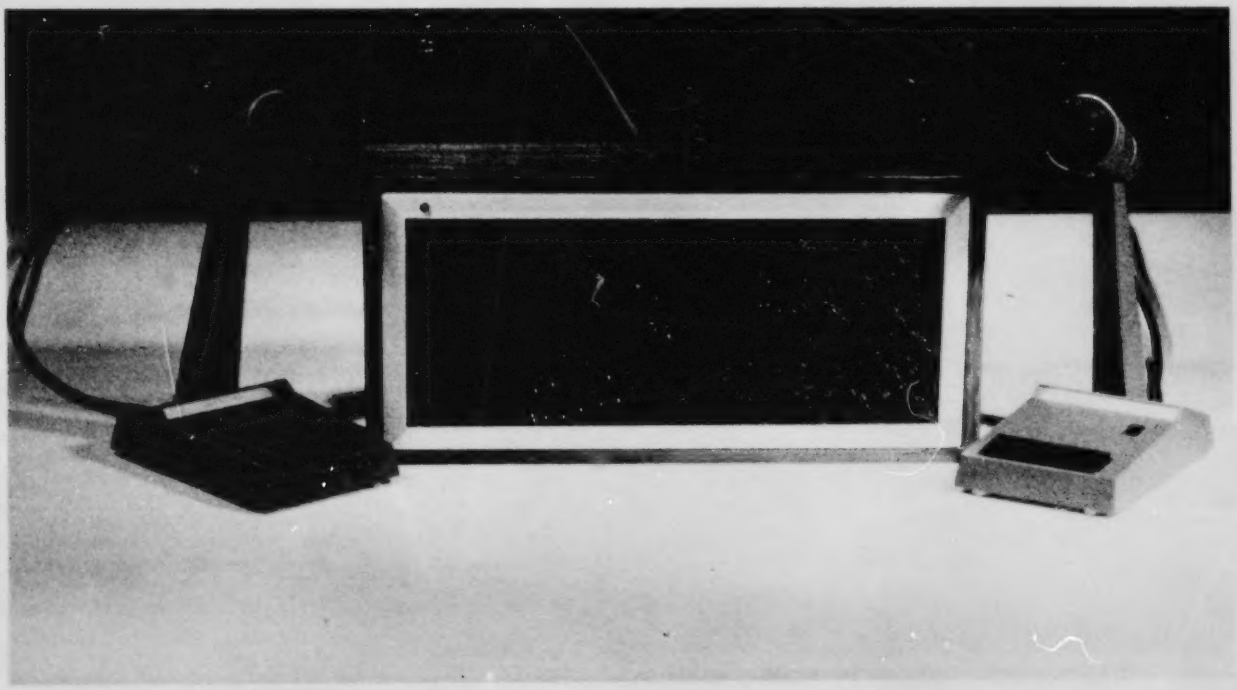
The POTS station equipment, the Darome Model 610, evolved from technical requirements substantially different than those governing the ETN station equipment. Portability was a primary consideration. ICS ships the unit, via common carrier, to an off-campus location for origination of an ETN program. At times, the equipment is hand carried on commercial aircraft. The Model 680 carrying case, which can fit under a commercial aircraft coach seat, houses the Model 610 convener, its basic 491 microphone and up to three additional Model 461 microphones.

Fundamentally, the 610 converts to a common 2-wire telephone circuit, rather than a 4-wire private circuit. Internally, the station receive channel and station transmit channel are very similar to their 4-wire counterparts. That is, the receive channel has a filter, preamplifier and 14 watt audio amplifier. Level adjustment is made by a single control. Transmission to ETN is with the same high quality press-to-activate cardioid microphone used with the Model 604. The difference lies in the switching. A voice operated switching circuit determines whether the station transmit or receive channel is connected to the phone line. In the quiescent mode both are off, and the channel on status is decided on a first come first serve basis.

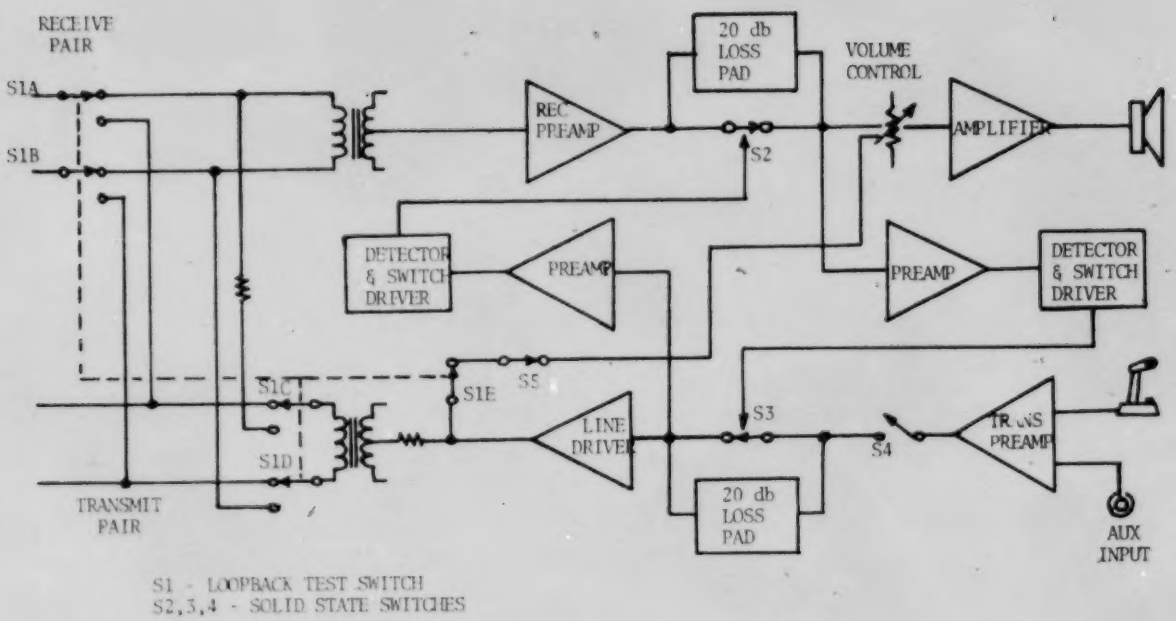
The Model 610 is operated in conjunction with a standard telephone, which is used to dial outgoing calls and provide a ringer for incoming calls. Transfer to the convener is made by a switch on the Model 491 microphone. A Light Emitting Diode (LED) provides an "on line" status indication. A short for the telephone line control leads is provided for telephone facilities with the hold circuit requirements. As with the Model 604, the addition of a Model 603 mixer allows the use of more microphones.

Design Constraints

In designing the ETN station equipment several noise constraints had to be considered; these can be broken into two categories, acoustic and electrical



MODEL 610



S1 - LOOPBACK TEST SWITCH
S2,3,4 - SOLID STATE SWITCHES

FUNCTIONAL BLOCK DIAGRAM MODEL 604 CONVENER

noise. Acoustic problems are related to the classroom size, the number of students in the classroom, the acoustical characteristics of the room, and the number of classrooms that are going to be tied together on the common network.

The classrooms used on the Educational Telephone Network vary in size from rooms seating 20 students to those seating 300, and sufficient acoustic power must be available to provide an intelligible sound level to any of these classrooms. A 14 watt RMS amplifier was deemed adequate, while three 5 1/2 inch speakers provide the sound projection. Additional speaker jacks on the basic unit facilitate adding column speakers to insure proper sound coverage in acoustically difficult rooms.

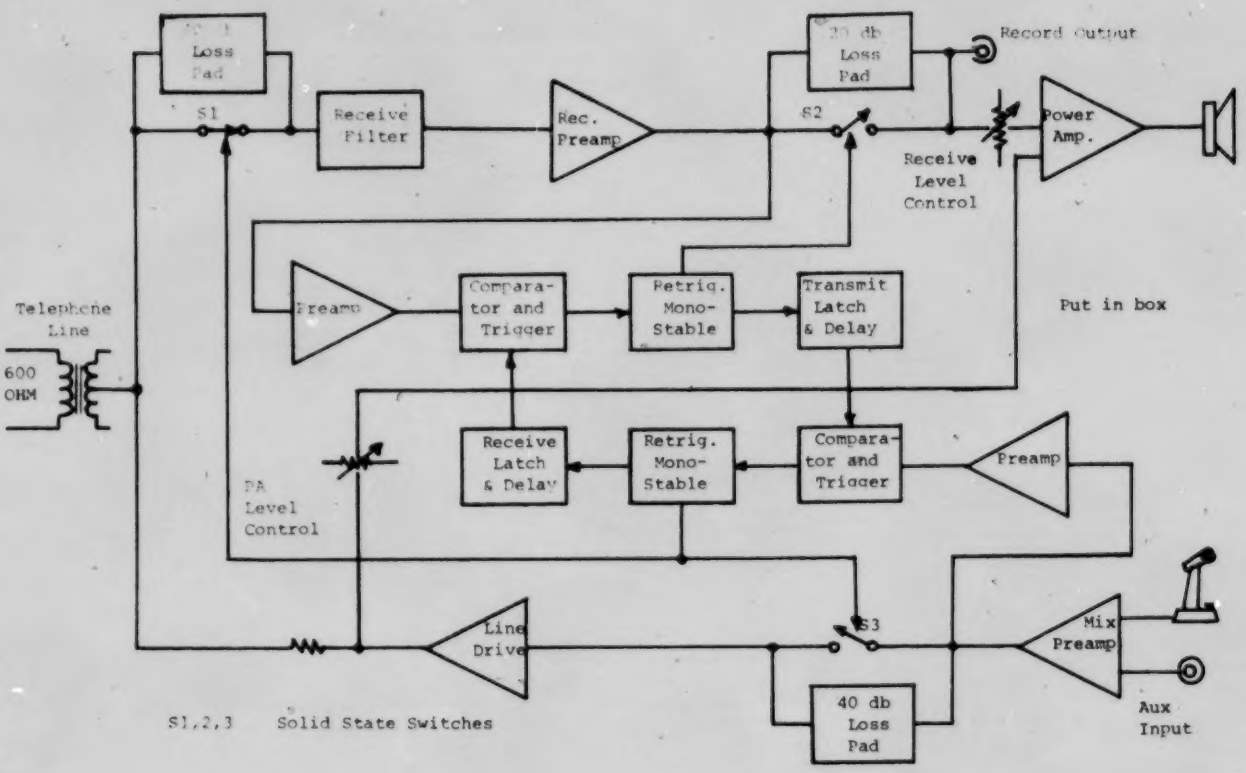
Classroom noise from numerous locations combining simultaneously on ETN would render the network useless for viable communications, therefore unused microphones in the classrooms are kept off to eliminate normal room noises. A voice-activated microphone was not chosen for this application simply because the noise to be eliminated would turn on the voice gate and pass the unwanted noise to the network. The microphone used with the Model 604 is a desk top mic with a built-in press-to-activate switch which requires minimum effort on the part of the student to initiate a question or response. (It was generally agreed that this would not significantly inhibit student response.)

The primary electrical noise constraint is the inherent noise generated by electronic devices, commonly referred to as "white noise." Again, with 200 such conference units tied on a common additive circuit, the amount of white noise would prohibit any intelligible communication. Since the student is required to press the drawbar on a microphone to speak, a solid state switch was added to shut-off the transmit circuit. This electronic switch, S4, was installed in such a way that the transmit circuit retains a proper termination of the phone line while prohibiting any electronic white noise from being transmitted to the telephone network. When the student presses the switch to speak, he activates both the microphone and the transmit circuit.

The noise constraints on the POTS station equipment are similar to those of the ETN model. Because of the generally smaller room sizes as well as the convener's smaller physical size, the Model 610 has one speaker rather than three. Acoustic output power is adequate for medium-sized meeting rooms.

For 2-wire, the voice activation, which was viewed as a disadvantage on the Model 604, is necessary to perform the transmit/receive switching on the Model 610. However, each microphone is disconnected from the transmit preamplifier until the drawbar is pressed. Once the microphone is connected, the voice switch (S3) operates when the input signal is of sufficient amplitude. Having the transmit in a normal off mode, again prevents the electronic white noise from adding to the network noise level.

The second category of restraints governing the design of the convener encompasses those affecting the signal-to-noise ratio. This ratio reflects the amount of signal present to the amount of noise present on the network at a given point and time. Conversely, with a given amount of noise on the network, a certain signal level must be maintained above that noise level to establish effective communications.



FUNCTIONAL BLOCK DIAGRAM - MODEL 610 CONVENER

Technical constraints to achieve a good signal-to-noise ratio involve all the elements in a sound reinforcement system, their location and relative locations within the room and the room acoustics. The room acoustics have to be accepted for two reasons. In many cases the classroom facilities are not under the control the University, such as those in hospitals, court houses and municipal libraries. The second reason is financial. The cost of acoustic treatment for 200 classrooms would be prohibitive.

The elements of the sound system include the microphone, amplifier and speaker. These elements must be optimized to eliminate two acoustic problems, commonly referred to as "echo" and "barrel effect." To the person at a network location listening to another person speaking from a second location, the listener experiences the illusion that the talker is at the bottom of a barrel.

This problem is aggrevated by live rooms. The live room has an abnormally high content of reflected sound because there are insufficient absorbtive surfaces to dissipate the acoustic energy. Consequently, the sound energy decays slowly, or the room is said to have a long reverberation time. The incident or direct sound is transmitted followed by transmissions of the reflected sound. The reflected sound is delayed from the incident sound and thus creates the barrel effect.

Basically, the barrel effect occurs in two situations. First, when the transmit preamplifier gain is increased so the speaker can be picked up across the room from the microphone. With the gain that high, the incident and reflected sounds will be transmitted creating the problem. The second occurs when the public address level is set too high. The talker's incident sound is picked up by the microphone and amplified to such a level that the reverberant sound from the sound system becomes significant relative to the talker's incident sound.

Minimizing these problems begins with selecting a highly directional cardioid microphone. The cardioid microphone responds well to signals directly on the front axis of the microphone and the response drops off sharply on the sides and rear of the microphone.

A second step towards a workable solution is to fix the transmit preamplifier gain. The fixed gain requires that the talker be within 12 to 18 inches of the front of the microphone to transmit an acceptable signal level. This eliminates the first cause of the barrel effect mentioned above. A benefit of this restriction is minimizing the transmission of room noises during the time a talker's microphone is on. To the remote network listener, the talker's voice is heard at a considerably higher level than the background noise within the room. Consequently, a good signal-to-noise ratio exists. In addition to limiting the transmit gain, the frequency response of the transmit preamplifier is tailored for speech intelligibility and telephone line compatibility.

To eliminate the second cause of the barrel effect, the public address level was fixed in relation to the receive level. As the receive is adjusted for a comfortable listening level, the public address level is generally maintained below the level at which the barrel effect begins. As a backup measure, a defeat switch eliminates the entire public address signal in an otherwise impossible acoustic situation.

The second acoustic problem to be eliminated is "echo," a common problem in telephone circuits. Echo is that portion of a transmitted signal which is retransmitted by the receiving end of the signal path. Generally, this signal is delayed in time by the transmission path. On ETN, echo occurs when a signal is received, amplified in the classroom, and transmitted back through ETN by an open microphone. Electronic delay is encountered in the telephone network and acoustic delay is encountered in the classroom. This echo is eliminated with the Model 604 by inserting 40 db of loss in each loop which is a potential source of echo, 20 db of loss in the talker's receive channel, and 20 db in the listener's transmit channel.

This is accomplished in the Darome Model 604 convener by the addition of voice operated switches S2 and S3. The switching operates as follows: with the microphone switch depressed and signal present on the microphone and transmit channel, a portion of the signal is fed to a preamplifier which in turn attenuates the receive channel by 20 db, that is, S2 opens. The figure of 20 db was arrived at by considering the number of stations involved, a typical number of microphones that might be in operation at any given time, and the amount of delay expected. This 20 db loss in the receive channel inhibits network signals from being amplified in the room and retransmitted on an open microphone. To further understand the operation, consider a unit at a station where the transmitted signal is received, that is, a listener's location. The listener's channel has a signal sensor on its input. With a signal present on the receive channel, the transmit channel is inhibited by 20 db (S3 opens) in the same way the receive channel is inhibited at the station where the signal is transmitted. In other words, if a student in a listening room were to press his microphone and attempt to speak with a signal present on his receive channel, he would be inhibited by 20 db.

Maintaining the signal-to-noise ratio on the POTS station equipment requires similar precautions, since the constraints are similar. That is, maintaining a sound reinforcement system in typically live rooms and obtaining a good transmit signal level. The transmit signal is picked up by the same cardioid microphone and amplified by a fixed gain preamplifier. The preamplifier response is tailored for speech.

The public address level is adjusted by a screwdriver control which is relatively inaccessible. The PA level is unaffected by the receive level control. Thus, the PA level can be set, and the sound system checked for stability prior to connection to the telephone facility. The wider signal level variations encountered on the DDD network dictate this control mechanization.

Echo on the POTS station equipment is eliminated by the simplex operation. The Model 610 is either in the transmit or receive mode, and therefore, cannot retransmit a station receive signal.

The simplex operation offers one signal-to-noise advantage. With the receive normally off (S2 open), telephone line noise below the gain threshold is not heard. The distraction of normal telephone line crosstalk is minimized since it can only be heard in the presence of a signal with sufficient amplitude to operate the voice switch. Therefore, the listener is not able to focus his attention on the background noise.

Maintenance

From the viewpoint of maintenance, the major constraint is the number of ETN units and their statewide distribution. The first question which arises after a station outage is, what has failed, the telephone line facility or the conference unit itself? The solution implemented in the design was to incorporate a loopback test feature. With the unit plugged into an ETN outlet, the loopback switch (S1) is placed in the test position. This switch ties the telephone line's transmit pair to its receive pair and internally disconnects the unit from the telephone line. A telephone company technician can then determine if the line is operational from the central office without dispatching service personnel to the classroom site.

As a second step, the conference unit itself can be tested. Activating the same loopback test switch also shorts the unit's transmit pair to its receive pair. By turning the receive to maximum volume and talking into a microphone, a signal is transmitted and received by the same unit. While the audio is distorted by the gating, it does verify the unit's operational integrity.

Maintenance on the conference unit itself is minimized by having all the internal electronic components on one plug-in printed circuit board. The customer's service department can replace the defective board with a spare and return the defective board to the manufacturer for repair. The university also has the option of replacing the entire defective unit.

Another potential source of maintenance on equipment is the operational complexity. The Model 604 controls have been minimized. There is no AC on/off switch; an LED indicates that the unit is plugged in and has power coming to it. A single control operates the station receive and public address levels. A PA on/off switch (S5) is provided, but is relatively inaccessible once the convener is set up.

A loopback test switch was mentioned above. The switch chosen was a spring return slide switch. This provides a simple operator presentation and maintains the unit in its normal mode of operation.

The POTS station equipment is a completely solid state design with integrated circuits for the majority of the circuit functions. For maintenance and test reasons, the circuitry is contained on a single plug-in printed circuit board. G 10 glass epoxy printed circuit material was chosen for strength against the abuse that portable equipment must endure.

A LED indicates when the Model 610 is actually on the telephone line. The unit left on is equivalent to an "off hook" condition on a normal telephone. The indicator serves as a reminder to turn the unit off after the completion of the conference.

Again, the controls were kept to a minimum; a

receive control and screwdriver adjustment for public address level. Only one microphone is equipped with a switch to answer the phone line, but all microphone jacks are the same, allowing this special microphone to be plugged into any of the four chassis jacks.

The Future of Teleconferencing Station Equipment

The requirements of ETN at the University of Wisconsin-Extension provided the stimulus for the design of new teleconferencing equipment. These requirements are not unique to a university's continuing education program; they apply to any institution or business which has a frequent need to communicate with groups of geographically dispersed individuals.

The United States Army is one example. From 25 regular Army schools with technical specialties, information and training must be provided to hundreds of regular Army bases and thousands of Army Reserve Centers. Numerous companies have sales offices throughout the states and around the world. Training and management efforts using teleconferencing could economically replace a significant percentage of the personal contact which occurs in these companies.

To enhance the use of teleconferencing, Darome developed other conveners which utilize slides as a visual aid. An instfuctor at one location can make a presentation over a conference network with each location having a slide projector and a duplicate set of slides. The instructor can then move all slide projectors forward and backward at his discretion.

Other improvements to existing Darome equipment which are in various stages of completion include the following:

--Built-in telephone line protection and repeat coils. Regulations governing the interconnection of customer-owned equipment to the telephone network have been challenged in recent years. As these regulations change and permit direct connection to the network, the inclusion of the protective devices will be mandatory.

--Adding built-in dial and ringer on the POTS station equipment. This would eliminate the need for a telephone, thereby simplifying the setup time and operation.

Areas of improvement under design investigation include the following:

--Smaller microphones. For portable equipment, the weight and size advantage would be substantial. For a Model 610 convener, the microphones and cords require twice the storage space occupied by the basic unit.

--Noise Mitigating Transmit Techniques. Noise on a telephone circuit not only detracts from the conference intelligibility, but also creates annoying gating problems. This is particularly evident on the POTS station equipment. Improvements in overall noise levels, while using standard telephone circuits, will certainly extend the use of teleconferencing throughout business and education.

--2-wire POTS station equipment which more closely resembles duplex operation. The present simplex operation of, say the Model 610, requires some discipline on the part of the conference participants. A more conversational unit would meet less opposition in marketplace.

All of the above would improve the state of the art in teleconferencing. However, substantial increases in the usage of teleconferencing will occur as it becomes as easy to use as the telephone itself. This means having customer operated conferencing equipment. Whether the conferencing bridge is on the customer's premise or is part of the telephone company central office should make little difference. The important feature will be to have the equipment available for use at the customer's option. Placing repeated conference calls could be done by a programmable dialer. The dialer could store the group of telephone numbers and originate the calls on demand or on a predetermined schedule.

The setup of the station equipment for short term usage could be greatly simplified by wireless microphones. The ability to pass microphones around the classroom or meeting room would facilitate more extensive participation. Other benefits which would occur are size and weight of the equipment.

Another development for the future is an individual student-type convener. The unit could be assigned to the student at the time of enrollment for a particular course.

For permanent installations in the "board room" category, a more sophisticated system is required. A modular, tabletop system which includes all microphones and speakers would circumvent an expensive custom installation. At the same time, it could provide a very usable teleconferencing system.

The telephone network is a natural teleconferencing vehicle. As special segments of educational requirements expand, a satellite could be used for nationwide programs. Closed circuit television could provide a mass medium for education. The student could not only obtain the video presentation, but through the use of an adapter, enter into two-way discussions with the instructor and other students.

The Distribution Network

The distribution network for ETN is leased from the Wisconsin Telephone Company. The network is provided as a private line talk, multipoint service. With the exception of the ICS studio control facility, the equipment, facilities, station terminations and portable station equipment are furnished by the Telephone Industry of the State of Wisconsin. The Wisconsin Telephone Company furnishes most of the backbone facilities as well as bridging and test locations. The Independent Telephone Companies furnish some of the facilities, such as bridging and station terminations. Portable station terminations are provided jointly by the Wisconsin Telephone Company and the General Telephone Company of Wisconsin. Many people and companies are involved in a coordinated effort by the Telephone Industry to provide a uniform grade of service throughout the State of Wisconsin.

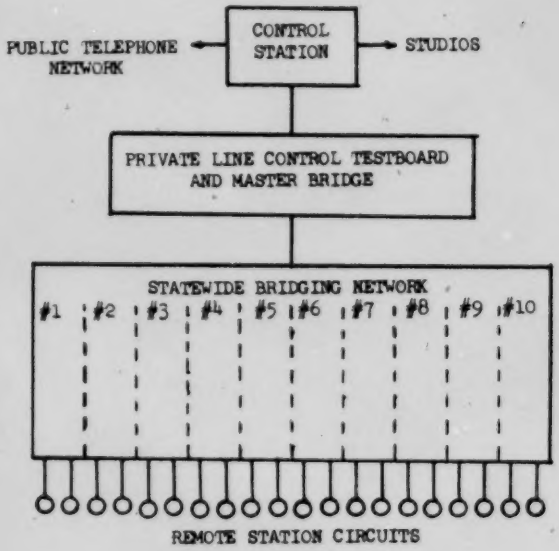
The Wisconsin Telephone Company coordinated the design and installation of the ETN. Uniform equipment

and facility specifications were agreed upon and standard station installation procedures were specified in detail. Taking these facts into consideration, this description of the network will not include the many junctions of ownership of facilities and variations of equipment due to ownership.

To start the description of the distribution network, consider the block diagram in Figure I as identifying the major elements.

FIGURE 1

BLOCK DIAGRAM OF MAJOR ELEMENTS OF ETN DISTRIBUTION NETWORK



Each major element along with its component parts will be described separately. The interconnection of the elements into a system will be described by identifying the interface between elements.

The Control Station

The control station is quite complex. The ICS concept of network administration and control dictates this complexity. Major components of the control station, identified in Figure 2, can best be described individually.

The Control Bridge

The heart of the control station is the 4-wire 6-way resistive bridge (Wescom 416). This bridge combines the three major inputs into a unified line feed for the ETN: the ICS studio control input is combined with the Public Telephone Network input and the input from remote studios or other networks. The output of the control station bridge feeds a line driving repeater.

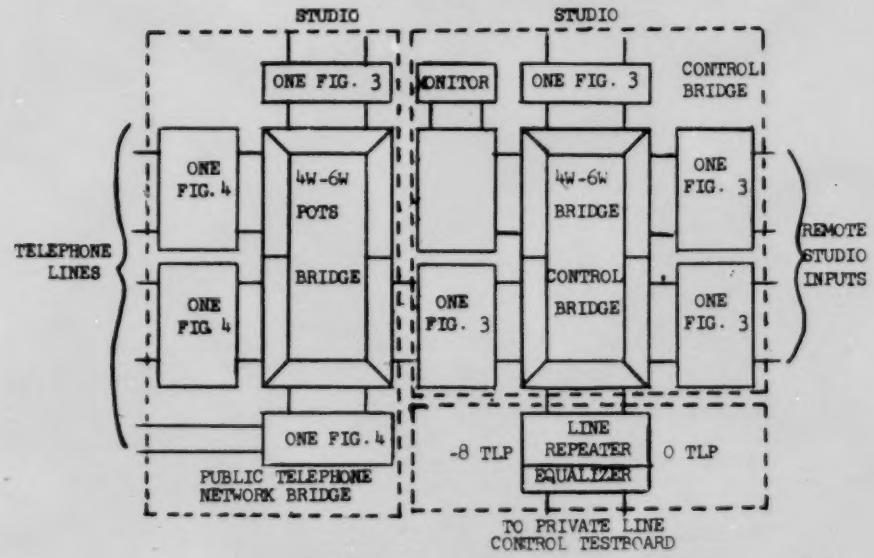
Each control bridge input is equipped with a Darome Convener switched gain repeater package (Figure 3), which is specially constructed for 19 inch relay rack mounting. This switched gain repeater provides the electrical equivalent of the portable remote station package. Each input to ETN, thereby, has the same switched gain characteristic.

The Public Telephone Network Bridge

The "Public Telephone Network Bridge" makes stations served by POTS available for interconnection to the ETN. Again the bridge used for this service is the 4-wire 6-way resistive bridge (Wescom 416). Each telephone line feeding the bridge is specially constructed to provide a 4-wire interface to the bridge. In addition the 4-wire design is extended to the serving telephone company central office switching machine.

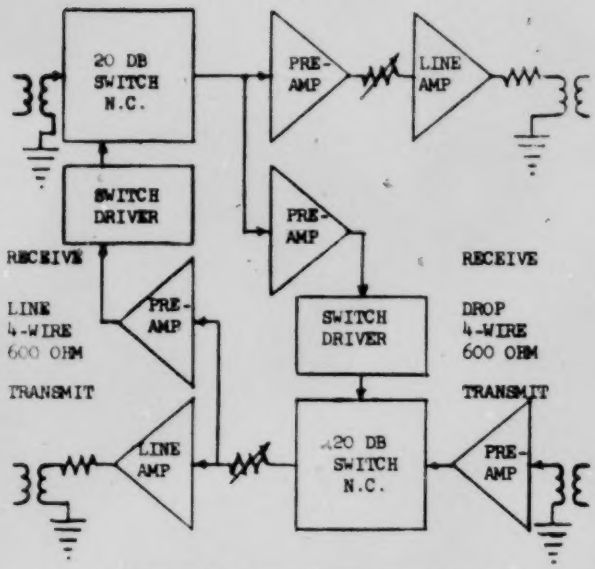
FIGURE 2

CONTROL STATION OF ETN DISTRIBUTION NETWORK



39

FIGURE 3



VOICE CROSS GATED REPEATER EQUIVALENT TO DAROME 604 CONVENER - 4-WIRE

At the 2-wire interface with the switching machine, each telephone line is equipped with a 24-type repeater. The central office transmit leg of each repeater is equipped with a low pass filter and a compression amplifier.

The filter serves to limit conditions which would create singing or echo at the 2-wire junction with POTS service. The compression amplifier serves as a level control gate for all signals entering ETN from the Public Telephone Network. Using the signal level from an intertoll telephone call as a reference, the amplifier provides full amplification to signals at or below this level range. Input signals that are higher in level are soft limited by the feedback circuitry of the amplifier.

The deviation of peak input signal levels from the reference level selected was observed to be -3 db to +10 db. The deviation of output signal levels for the same observed sample was -3 db to +3 db. This range appears satisfactory for interconnection to the private line network. Since the signals being limited generally have a high signal-to-noise ratio, little noticeable signal to noise ratio degradation may be attributed to the compression amplifiers. Conditioned in this manner, a program from the public telephone network is compatible with the private line characteristics and program of ETN.

The Line Repeater

The line drive repeater connects the control bridge to the local cable pairs feeding the Plant Control Testboard. In so doing it sets the transmit

FIGURE 5

COMPRESSION AMPLIFIER OUTPUT SIGNAL LEVEL DEVIATION AS A FUNCTION OF INPUT SIGNAL LEVEL DEVIATION

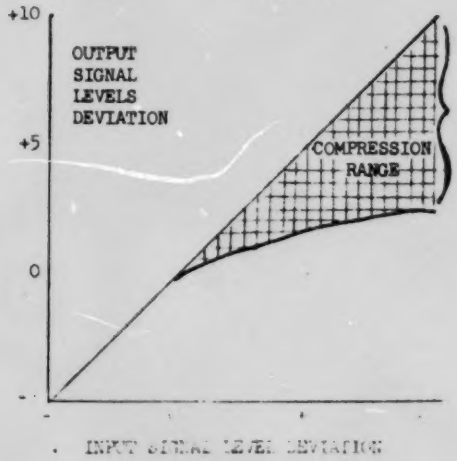
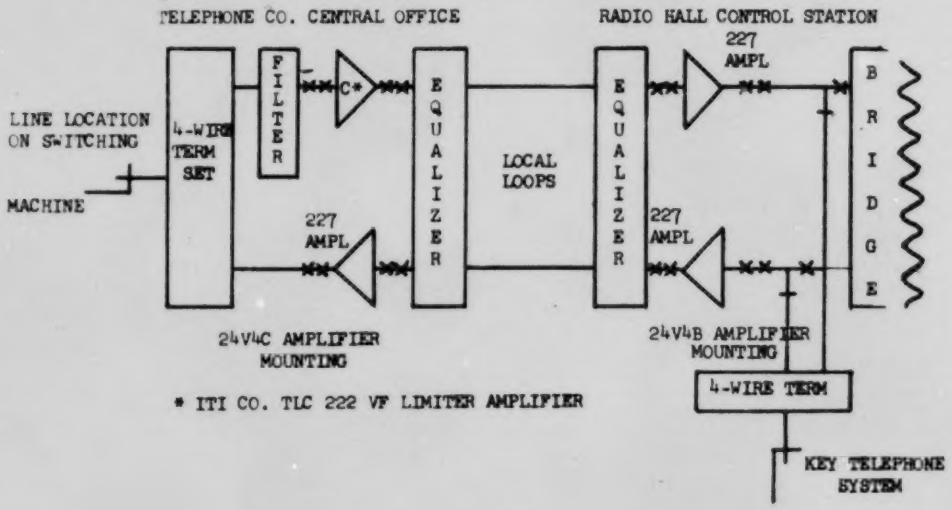


FIGURE 4

4-WIRE BRIDGE TELEPHONE LINE



* ITI CO. TLC 222 VF LIMITER AMPLIFIER

level to the line at 0 db Transmission Level Point (TLP), and the receive level from the line at -8 db TLP. These line interface levels are also used at each remote station interface throughout the network. The limiting signal level at the line input is 0 VU.

Normally a telephone instrument oriented system design would use -16 db TLP receive level. This signal level causes the telephone instruments to deliver the correct amount of acoustic power to the ear. The design deviation to -8 db could be made because telephone instruments are not used on ETN. Zero dbm receive level was not used since this level would have required active amplifiers at remote stations. The change to -8 dbm provided added signal-to-noise ratio margin for ETN.

The final function of the line drive repeater is to provide line equalization to improve band pass characteristics of the cable pairs between the Private Line Testboard and the Control Station.

The Private Line Control Testboard and Master Bridge

For ETN, the domain of the Madison Private Line Control Testboard (PLCTB) includes all telephone industry provided equipment and facilities used on the network. This responsibility is shared with regional private line serving test centers. For the purpose of this description, only equipment and facilities terminating on the Madison Testboard and the master bridge will be considered.

The cable pairs connecting the control station to the Private Line Control Testboard are accessible at the testboard normal patch jacks. The drop side of these jacks feeds the master bridge of ETN through a line equalizer and repeater. The Master Bridge interconnects the control station through region control circuits to the ten regional distribution networks. This bridge does not connect directly to remote stations.

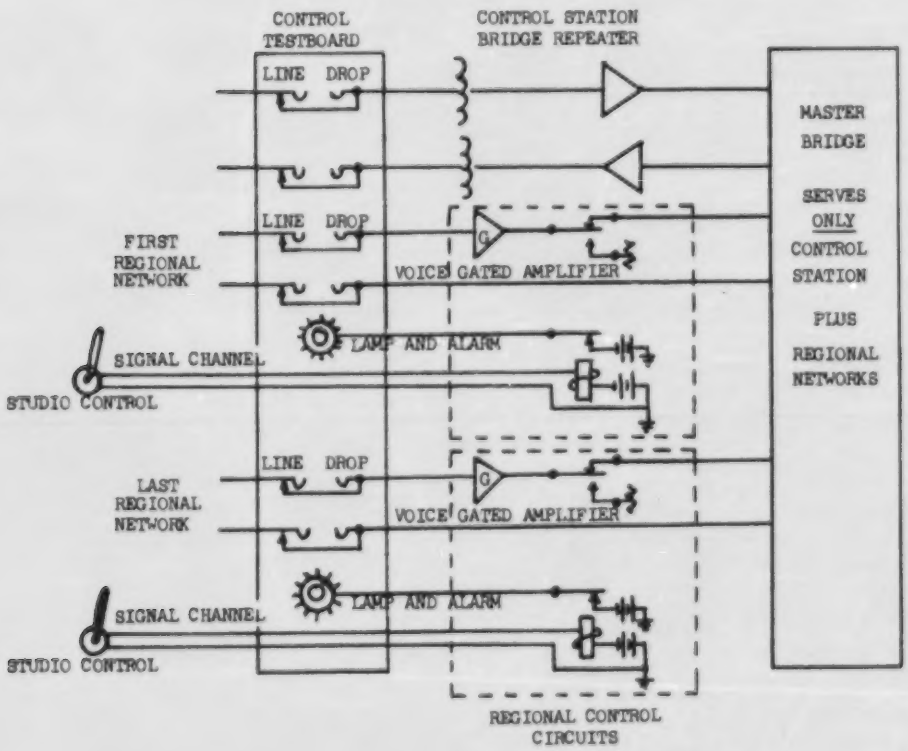
Each regional control circuit is connected by a signal channel to the Control Console at Studio Control for ETN. Each signal circuit operates a relay which 1) opens and terminates the receive signal path from the regional associated, and 2) signals the control testboard by operating a testboard lamp and alarm circuit. This function serves to isolate a regional master bridge receive port from the rest of ETN. (Trouble in one region does not affect other regions.) It also serves to report a regional trouble condition immediately to the PLCTB.

In its normal state the Master Bridge serves as a one-way distribution network passing program from the control station to each regional network. The voice-gated amplifier in each regional Master Bridge receive leg is blocking regional noise from the Master Bridge. This is the condition with all remote stations listening to program. In this state, ETN is a broadcast-only conference circuit.

When a remote station addresses the ETN, the associated regional voice-gated amplifier gate triggers, transmitting the signal to the Master Bridge

FIGURE 6

MASTER BRIDGE AND PRIVATE LINE CONTROL TESTBOARD



41

and on to the other regions. All stations receive the signal emanating from the remote station. In this state, ETN is a two-way conference circuit.

In the same manner, if a carrier channel or cable pair within a region were to fail, the noise generated would also trigger the voice-gated amplifier. This would, in turn, disable the entire ETN. The Studio Control console, by operating one control circuit relay at a time, can isolate the offending region in seconds, thus restoring nine of ten regions to service. The control testboard then swings into action, alerted by the operated lamp and alarm circuit associated with the offending region. They immediately contact the regional serving testboard involved. Together they further isolate the regional problem to a Primary Bridge leg. The offending leg is isolated from the primary bridge. The region, less the offending stations, is restored to service within minutes of the failure. Normal trouble clearance procedures are then applied to the offending stations.

The control testboard also receives trouble reports directly from remote stations when they experience difficulty. All customer trouble reports are made directly to the Madison PLCTB by ETN users.

One additional tool serves the PLCTB. A monitor leg from the Master Bridge is terminated in a portable speaker at the board. This serves the testman by providing immediate access to monitor trouble on the network without interrupting programs in progress. It also serves as a quick test for early hour testboard coverage. Trouble indications that appeared during the uncovered night hours can be investigated before the first ETN program in the morning.

The Statewide Bridging Networks

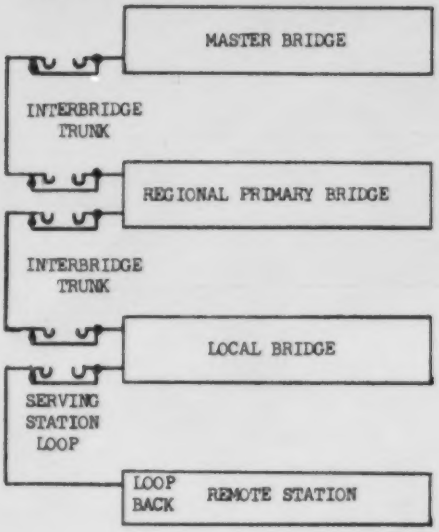
The practical size limit for multipoint private line telephone networks is twenty stations. The noise level, created by the multiple facilities and circuit equipment terminations connected together, is usually the limiting factor. ETN, with 200 ports, is able to provide service that satisfies users because the network is constructed from ten noise-isolated regions. Each region of the network serves 20 stations. Regions are isolated by the voice-gated amplifier at the Master Bridge regional receive port. The regional arrangement limits the noise level experienced at a remote station to noise generated by the facilities and terminations within a region.

When a remote station addresses the ETN, the voice-gate isolating its associated region from the Master Bridge opens. This allows the signal along with its accompanying noise level to enter the Master Bridge. The remote station signal and its accompanying regional network level is fed through to the other regional networks. The combining of this signal and noise with the steady state noise in the other regional networks raises the noise level in other regions. Under normal operating circumstances this is the highest noise level expected within a regional bridging network. (Equivalent noise to a station on a 40 port network for the period of conversation from remote station plus 100 milliseconds.)

Network Limits. Each region is served by one primary bridge which in turn serves remote stations either directly or through local bridges. The maximum number of tandem interbridge trunks within a region is two (Figure 7). On an interregional basis, this limits the number of tandem links in the transmission path between remote stations to four mid links (between bridges) and two end links.

FIGURE 7

TANDEM LINKS - MAXIMUM NUMBER
FROM MASTER BRIDGE TO REMOTE STATION



Two point private line talk service is designed to meet an overall frequency attenuation distortion requirement as listed in Table 1. Multipoint service requires the same overall result. To attain this result each link is given a special consideration requirement also listed in Table 1. Each link is measured and maintained separately. Overall results are not measured for installation or maintenance purposes.

TABLE 1

PRIVATE LINE BANDWIDTH CIRCUIT ATTENUATION DISTORTION REQUIREMENTS (DB)*

FREQUENCY RANGE-HZ	OVERALL 2-POINT	MULTIPOINT END LINK	-4 MID LINK MID LINK
500 - 2500	-2 to +8	-0.8 to +3.5	-0.8 to +3.0
300 - 3000	-3 to +12	-1.5 to +4.5	-1.0 to +4.5

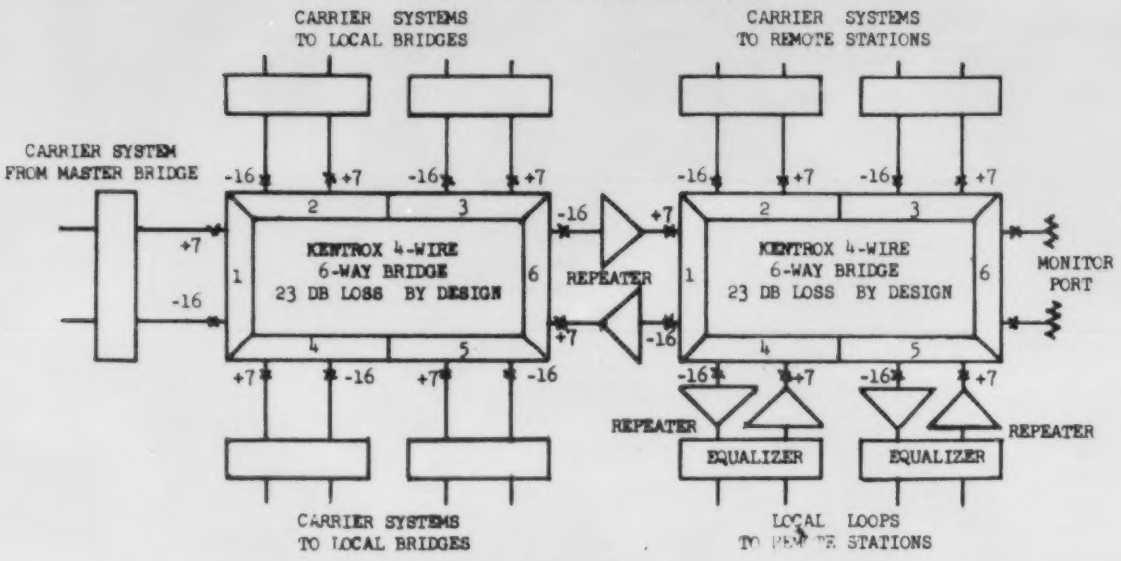
*DEVIATION FROM 1000 HZ LOSS
(-) LESS LOSS
(+) MORE LOSS

Network Equipment

The equipment used to provide the distributing network bridging arrangements consists of testboard normal jacks, 4-wire repeaters and Kentrox 4-wire - 6-way bridges. Carrier terminals are interconnected directly to the 46-type bridges which have 23 db of loss. The bridge serves in place of 23 db pads between interconnected carrier channels reducing wiring and equipment components necessary at bridging points. A typical regional bridge interconnection is

FIGURE 8

TYPICAL REGIONAL BRIDGE



block diagramed in Figure 8.

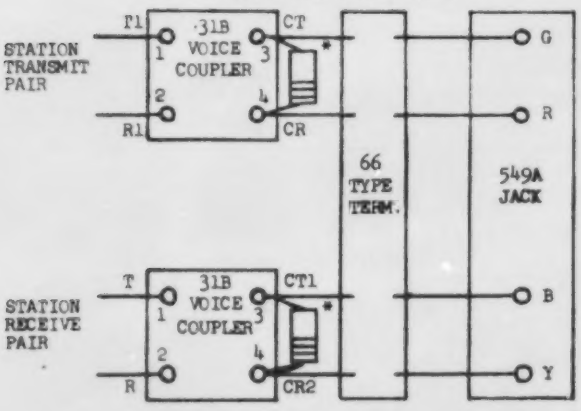
The Remote Station Circuit

Each remote station is served from a regional, primary or local bridge. The facilities involved include interoffice cable or carrier, in addition to local loops. Circuit equipment is added to the station circuit to provide signal level treatment, as well as frequency attenuation and bandwidth equalization. The hand-wired station equipment provides a standard 4-wire termination and wiring to 4-wire telephone jacks.

The remote station circuit is of modular design. Two standard modules have been established. The first type is used when the serving bridge port location is other than the serving central office for the local loop to the station (Figure 10). The second type is used when the serving bridge port and the station loop terminations are in the same wire center or central office.

FIGURE 9

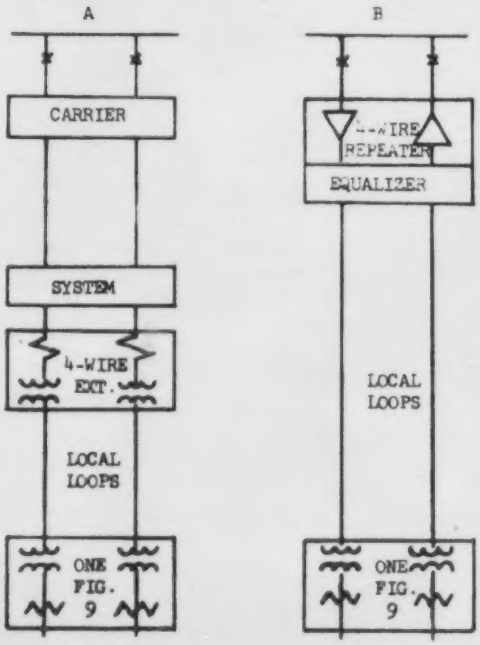
REMOTE STATION HARD WIRED
TERMINATING ARRANGEMENT



*1500 OHM 1 WATT RESISTOR SERVES AS PARTIAL TERMINATION FOR STATION WHEN PORTABLE EQUIPMENT IS NOT CONNECTED TO JACKS.

FIGURE 10

REMOTE STATION CIRCUIT MODULES



Using The Transmission System To Overcome Network Obstacles and Constraints

The difference between a mediocre transmission system and a superb transmission system is fine tuning. In the case of ETN, the transmission system appears ordinary enough, but there is evidence of fine tuning. Starting at the control station and proceeding through the system, fine tuning is accomplished by these specific details:

1. The unique blending of POTS and studio programming using the ITI limited amplifier to equalize signals levels.
2. The uniform switched-gain repeater characteristic of signal treatment for all network ports.
3. Optimizing the network line receive level to -8.0 db TLP.
4. The Master Bridge concept to combine regional networks.
5. The regional control circuits utilizing the ITI 1130 voice-gated amplifier to isolate regional noise level.
6. The relay switching within the regional control circuit for customer initiated maintenance action.
7. The two interbridge link network design for improved frequency attenuation performance as well as rapid trouble isolation. (Overall design is four links but studio to remote station is two links.)
8. Uniform bridging configuration and equipment to minimize design and maintenance practices.
9. Passive hard-wired station terminations.
10. Uniform station installation instructions for Wisconsin Telephone Company stations as well as Independent Telephone Company stations.
11. Uniform Testing instructions distributed with Special Service Work Order to each installer and testman.
12. One trouble report center for all stations on network.
13. Station loopback feature provided by portable station package to provide speed testing.
14. The usage of specialized equipment to hand tailor the network.
    - A) TLC 222 VF limited amplifier[(a)]
    - B) TLC 1130-1 voice-gated amplifier[(a)]
    - C) Kentrox 4-wire 6-way distributing frame mounted bridge with 23 db loss design.[(b)]
    - D) Darome switched gain repeater at each ETN station port.[(c)]

(a) ITI Electronics, Inc., 369 Levington Avenue, Clifton, N.J. 07015

(b) Kentrox Industries, Inc., 14335 N.W. Science Park Drive, Portland, Oregon 97229

(c) Darome, 711 East Diggins Street, Harvard, Illinois 60033

Since system design must be user sensitive, each of these fine-tuning features was introduced into the transmission system design as improvements were deemed necessary. As feedback from the user indicates a weak spot, a solution eliminating the weak spot is required. This network design is not constrained by hardware, Telephone Company, University, or even normal maintenance practices. History of growth has demonstrated ETN to be truly user sensitive. Each network improvement feature is instigated by user feedback.

Future Speculation

There is no time to be complacent in a system which plans to meet user requirements: Continuing feedback indicates that the user is always reaching to improve his lot. Where can the transmission system be improved to meet this challenge?

Improve Low Station Usage Time. The first apparent ETN weak spot is low individual station usage time. Although the network usage time is fully committed, there are many stations idle for any particular program. ETN could be more effective if small groups of stations could be switched into POTS bridging arrangements. These POTS bridges could then provide additional program capacity for the stations so connected. It is conceivable that regional POTS bridges would meet this need.

If a POTS bridge were established, it could be used in reverse to bridge POTS stations to ETN at Regional Bridges. This would provide programs to stations which are not now economical because of low usage. This bridging could be best accomplished on a meet-me basis of switching (remote station controls access to network).

Privacy Feature For Control of Stations Receiving Program. There is always the possibility that stations not contracting for a program could bootleg the program. This is the normal state of the network. Each station has access to the network at all times. A privacy feature, somewhat simplified from SS3 signaling system now offered for multipoint private line systems, could be developed to channelize ETN for major station groups, e.g., hospitals, libraries, court houses, University of Wisconsin-Extension Centers.

User Feedback Requirements. Future changes in this Telephone Industry provided transmission system will be dictated by the feedback from user surveys. At this point, it is difficult to predict the developments this feedback may require. Suffice it to say, every effort will be made to expand the educational opportunities offered through this combined University and Telephone Industry offering.

Studio Control Complex

The ETN studio origination and control complex at Old Radio Hall on the University of Wisconsin-Madison campus, is equipped to provide operational control of the network. Within the complex are studios from which instructors broadcast their programs, and the control room, which is manned by a trained engineering technician who handles all equipment and adjustments during each program.

The instructor has a minimum of technology to cope with. During his presentation, he need only speak into a microphone, and he can hear a comment or question from the network through a loudspeaker. Unlike a regular ETN station microphone, there is no push-to-activate switch. All controls are handled by a trained engineer.

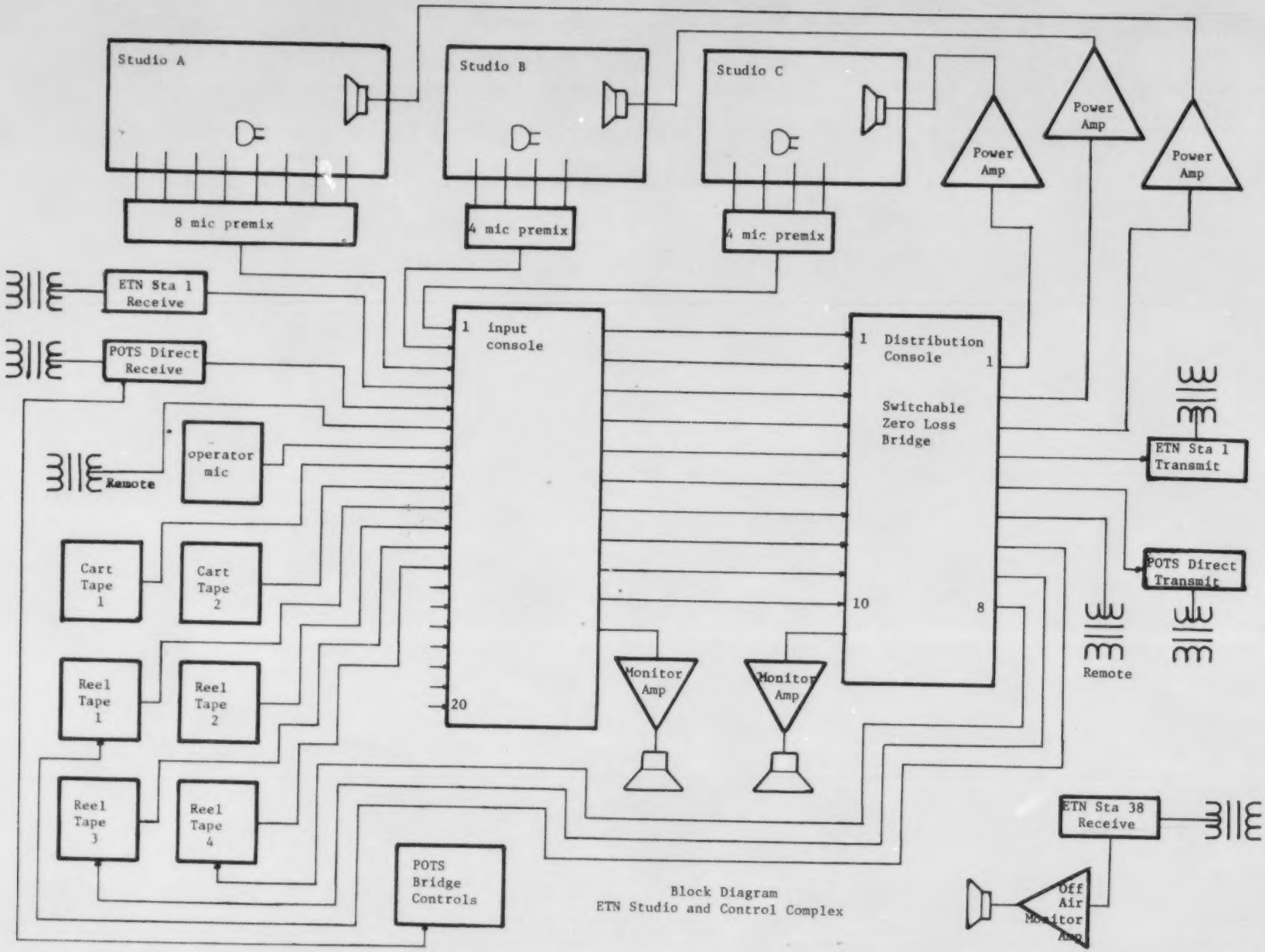
The studios and control room include the necessary equipment for use by the operator to guarantee a smooth trouble free program. Because the operator runs all technical aspects of the program, the instructor is free to concentrate his efforts on his material and presentation. The operator's role can be illustrated most clearly by a description of the steps involved in setting up and operating a typical program.

Before a program, the operator refers to a daily schedule and program format book, similar to a radio broadcast conformity sheet, which outlines the particular technical and operational requirements for that program. The studio is arranged with enough mics and chairs for participants, and, if slides or visuals are to be used, an appropriate projector and screen are set up. Insert tapes are loaded on tape machines and another tape machine is used to record the entire program.

The switches and controls on the input and distribution consoles are arranged in accordance with operational requirements. When the instructor arrives, the operator greets him and reviews the program format to check for any changes. If a participant is to be connected to ETN via POTS, the operator places the call, gets the party on the line and, when ready, connects him to ETN. A program introduction on cartridge tape is played, the studio microphones are turned on and the instructor begins.

During the program the operator plays the tape inserts on cue from the instructor. If a problem develops with the control room equipment, he quickly transfers to a back up unit. If he hears trouble on the network, such as noise, etc., he uses controls to isolate the offending region and reports it to the

ETN CONTROL ROOM



Block Diagram
ETN Studio and Control Complex

telephone test board which then helps to isolate and correct the problem. Most of the time problems do not occur, however the operator continuously monitors the program operation in case of trouble. When it does happen, all efforts are made to minimize the effect on the program.

The ETN control room is centrally located, with studios A, B and C adjoining on three sides. The operator and instructor have visual contact through large windows. The studios and control room are sound proof, and access to the rooms is provided through connecting doors and soundlocks.

Studios are furnished with conference style tables surrounded by comfortable padded swivel chairs. Studios B and C accommodate up to 8 people, while Studio A is much larger and can hold class sizes of about 40 students. Various table and chair arrangements are used depending on class size, although the smaller studios are generally left set up in the conference table arrangement.

The studios are treated to minimize reverberation and are essentially dead acoustically. Four high quality cardioid microphones give proper coverage in Studios B and C respectively, while up to eight are used in Studio A. In very large classes, no attempt is made to give a mic to each student.

ETN ORIGINATION CENTER

Generally, they share a single microphone provided at each table.

The microphones are assigned to each studio and are terminated in a premix amplifier which is mounted in the control room. Here the operator can control the gain of each mic path. The outputs of the premix amplifiers are balanced line level and appear as an audio source at the input console.

Audio Tape Machine

Professional audio quality tape recorder/players are used in the ICS operation. Though the wide frequency response and low noise figures are not needed on ETN with its limited band width and inherent noise characteristics, they were chosen for other qualities: long term reliability, serviceability and ease of operation. Ampex AG350 has been in continuous use for ten years. Equipment includes four reel-to-reel machines and two broadcast-type cartridge machines which are employed to insert introductions, logos and other short segments used regularly in programs.

A standard tape-and-track format has been established. Seven inch reel tapes of either 1.5 mil or 1.0 mil A-V grade are recorded or played at either 3.75 or 7.5 ips 1/2 track mono. (Generally, 3.75 ips tape speed is used). Insert tapes are previewed and processed to adjust any deviations to these standards if needed. Maintaining these standards make it possible to quickly provide tape copies of the programs in either reel-to-reel or cassette, using high speed duplicating equipment. The output of the tape machines appears as audio sources at the input console.

The cassette format is used in the control room on occasions, generally as playback of an insert. Until recently cassette machines were not sufficiently reliable to be used for program mastering.

Other Audio Sources

A control room microphone allows the operator to talk to any point on the network. This mic can be used when assistance is required or if a trouble condition is to be cleared. Remote lines, provided to a number of services and class rooms outside Old Radio Hall, are used when programs originate from these points.

The ETN Station #1 receive port (the studio receive) appears in the control room and is the interface with the telephone company bridging equipment. The POTS bridging and conferencing features can be used independently of ETN when recording a program insert of a conference call with a studio group. This appears as the POTS (DDD) receive direct, and is an interface with telephone company provided equipment. These sources also appear at the input console.

Input Console

The input console, a modified radio broadcast control board, is used for a number of its basic features. It is the appearance point for all the audio sources. Input circuitry provides the necessary impedence matching and signal amplification or attenuation needed to properly terminate the varied audio source equipment. The console contains ten input channels with fader control of source levels and A or B source selector switch which doubles the sources readily available to twenty.

A program-audition select switch connects the audio source directly to the distribution console or to an audition monitor and amplifier and VU meter. This feature allows the operator to cue and listen to any of the audio sources without connecting them into the network or program.

Distribution Console

When a channel is switched to program at the input console, it is tied directly to its respective input at the distribution console. The distribution console is a switchable, zero loss bridge with ten inputs and eight outputs. Any combination of inputs can be tied to any combination of outputs. Isolation greater than 80 db is maintained between the inputs. Each of the outputs has a level control and on-off switch. The output amplifiers will drive a 600 ohm line to +14 dbm, enough power to meet most interface requirements. The outputs are tied to various points in the system and each feed point receives a combination of all other signals except its own new source. As an example, the speaker mounted in the studio is powered by an amplifier which is fed by one of the distribution console outputs. With the proper selection of source switching, the studio speaker receives all other signal sources, but not the signals from the studio microphones. This feature is needed to maintain the necessary 4-wire isolation integrity to be compatible with the ETN transmission system. Outputs are assigned to tape machines for recording, to ETN station #1 transmit to tie the studios to the rest of the network, to POTS direct and to other studio monitors.

Monitoring

Monitoring functions are an important tool for the control room operator. To successfully isolate trouble, he must be able to monitor the signals at each stage in the process.

In addition to the audition monitoring mentioned earlier, the operator is able to monitor any of the eight distribution console outputs. The program monitor amplifier and VU meter can be selectively switched to any of the outputs without loading or affecting it. Thus, if a low level report is received, the operator can check the outgoing level quickly and correct it if necessary.

Another monitoring feature is similar to a broadcast off-the-air monitor. A sample of the receive signal for ETN Station #38 is fed to a monitor amplifier speaker and a VU meter. Station #38 appears in Old Radio Hall. A proper signal appearing at this monitor point assures the control operator that Station #1 transmit signals are leaving the building and getting through the master ETN bridges at the telephone company.

Patch Panel

Signal runs between equipment are routed through a patch panel. For example, the output of a tape machine appears on the tips of a patch port. The input to Channel 3A of the input console appears on the tips of another patch port. If the output of the tape machine is to be assigned to input Channel 3A, the tips are connected together through the normals. Connections between the input and distribution consoles also appear at the patch panel in this manner. A patch cord plugged into one of the patch ports breaks the normal connection. This feature provides versatility in equipment interconnection arrangements and also facilities isolating or bypassing faulty equipment. The patch panel provides a termination point for numerous sources and feeds that are used only occasionally and, therefore, not normalled to an input or output.

POTS Bridge Controls

The POTS to ETN bridge controls are presented to the operator as shown in the picture. There are two independent sets of three lines each which appear operationally as a standard six button telephone. The third set shown is the ICS general telephone extension. In this case the push buttons are mounted on a panel and connected to a trimline telephone set. The control action is push-push with latching relays controlling the circuitry. The buttons are color coded, and lighted to indicate status. The controls permit versatile use of the lines available. Each set of lines can be used for a conference call and both sets can be bridged together providing a six party conference. Each set of lines can be used directly with the studio or tied to ETN. As an example, one set can connect up to three callers to ETN, while the other set is being used to record a conference call interview as an insert for a future program. This versatility allows ICS to make maximum use of the service.

Constraints

Consideration for signal-to-noise ratio is important here as well as with other parts of the system. Noise from outside the studio is minimized by using soundproofing techniques and noise generated within the studio is minimized by using highly directional microphones. Interaction between the studio speaker and the microphone must be kept to a minimum. The problem is the same as discussed earlier in relation to station equipment.

The gain and number of simultaneously open studio microphones is limited by the presence of the live monitor speaker. Too high a speaker level or microphone gain creates echo conditions. These echo signals can cause gating problems which result in interruption of the program.

Some increase in the range of microphone gain is accomplished by acoustically treating the studios. By making the studio acoustically dead, better advantage can be made of the directional properties of the various elements involved.

The control room evolved over a period of time along with the development of the rest of the network. Control consoles and equipment to meet the unique needs of the ETN control function were not commercially available, at least not at a practical price. The solution was to modify some equipment from the broadcast industry design and build the rest in-house.

Rapid growth and changing demands require a high degree of versatility in control room equipment. As more features are added, an increase in operational complexity results. It is important, therefore, to review the control operation periodically to reduce this complexity. Redundancies in controls can be eliminated and those features regularly used can be locked in a set of operational patterns. Effort spent in planning the console layout and control inter-relationships will pay off in a smooth operation and increased operator effectiveness. Electronic white noise and cross talk generated by control room equipment are minimized by utilizing proper design and installation techniques.

Future

As the use of teleconferencing in education grows, so will the need to maximize the use of available telecommunication facilities. ETN, instead of being a huge party line, will be broken into a number of smaller regional networks which can be programmed separately or tied together as program distribution patterns require. The trend toward smaller networks using POTS and on premise conferencing equipment creates another need for a technical supervisory control function.

A number of current functions and new features can best be controlled by computer:

--As part of the control complex the computer can automatically set up network arrangements as identified by enrollments.

--Networks using POTS can be automatically dialed and conferenced.

--Continuous circuit testing can be carried out and trouble shooting streamlined.

--Visual support equipment at all locations, such as a slide projector and video tape machines, can be simultaneously controlled from the origination point.

The computer can poll and register non-verbal responses from each station location. For example students could take a multiple choice exam in real time, with the results immediately tabulated.

One way of measuring the acceptance of program technical aspects is by student user assessment. The technical components of ETN and SEEN programs, along with several other areas, are systematically evaluated through the use of the Program Evaluation Checklist (PEC).

Students in a random sample of ETN and SEEN programs are asked to rate each of the statements on the PEC and determine how well the positive statement fits the description of the program. On a 0 to 9 scale, 0 represents minimum approval while 9 denotes maximum acceptance.

In 1975, after the new Darome Edu-Com II equipment had been in use for two semesters, a random sample among 632 participants gave ETN's technical components a mean rating of 8.45. With this degree of user approval, the future of teleconferencing and its associated supervisory control function is only limited by the creativity and ability of educators and engineers to adapt technology to serve outreach education efforts.

AUDIO TELECONFERENCING

TRANSMISSION SYSTEMS & CONSIDERATIONS

INDEX

INTRODUCTION -

A brief definition; purpose of the paper and how it is organized. 1

OVERVIEW -

A brief discussion of audio teleconferencing arrangements; modes of operation; methods of establishing teleconferences; and interfaces. 1

TRANSMISSION SYSTEMS -

Types of systems; facilities; and bridging methods. 5

TRANSMISSION CONSIDERATIONS -

A discussion of some parameters effecting transmission; impact of the parameters on Audio Teleconferencing Systems; noise and echo treatment; and bridging elements. 7

CONCLUSIONS -

Some recommendations and a glimpse at the future. 13

AUDIO TELECONFERENCING
TRANSMISSION SYSTEMS & CONSIDERATIONS

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INTRODUCTION

A brief definition; purpose of the paper and how it is organized

Definition

Since this paper deals primarily with the transmission systems and considerations it seems fitting that the opening paragraphs should at least contain a brief definition of audio teleconferencing. Thus, in its simplest form let us consider audio teleconferencing to be an interaction of one individual, the conference originator, with two or more individuals, the conferees. Let us further state that such interaction be accomplished by some form of wire link such as a dedicated facility or a connection established over one or more switched telecommunication networks.

Audio teleconferencing may also be considered as the interconnection of two or more groups or assemblies by wire link allowing the simultaneous interchange of ideas of each conference. Finally in its more complex form audio teleconferencing may involve the interconnection by communication links of several individuals and groups interacting for a common cause.

Purpose

My main purpose however, is to present a discussion of the various transmission systems used to achieve audio teleconferencing. In addition it is my intention to address certain transmission parameters and considerations for controlling them in practical audio teleconferencing systems. It is not my intention to provide a mathematical analysis of the parameters but instead to show how each impacts quantitatively on the final system design, and what steps may be taken to control each.

Paper Organization

The paper is divided into four main sections. The first is a brief overview of audio teleconferencing. Readers already acquainted with audio teleconferencing systems may wish to proceed directly to the second section which deals more specifically with transmission systems. The third section deals with transmission considerations. Finally the paper concludes with some recommendations and a view of future audio teleconferencing trends.

OVERVIEW

A brief discussion of audio teleconferencing arrangements; modes of operation; methods of establishing teleconferences; and interfaces.

Conferencing Arrangements

The interconnection of three or more individuals by wire and/or wireless communications links is by no means a new concept. Multipoint private or dedicated networks have been around since the earliest days of telegraphy as well as telephony. Books have been written concerning such arrangements. The interconnection of assemblies of people through the marriage of specially treated conference rooms is a somewhat new concept, coming into focus more in keeping with the computer and space ages and fostered more recently by the energy crisis.

As stated in the introduction it is not the intention of this paper to dwell on the details of audio teleconferencing arrangements but rather to briefly describe some of the possibilities more as background for the uninitiated reader so as to make the subsequent discussion of transus by means of a series of scenarios review a few of the configurations. Methods of establishing audio teleconferences and how they are run, i.e. mode of operation, will be ignored for the time being but will be discussed later.

Room to Room. Mr. "A", an executive of Sales Promotions Inc. requests a meeting of his immediate subordinates to discuss a new product promotion scheme. Mr. "A" and staff meet in a conference room in City A. Mr. "A" and his staff wish to include in their discussion a Graphic Arts Company located at City B. Mr. "B" of Graphic Inc. assembles his staff in their conference room. The two rooms are linked together by a suitable communications link as pictured in Figure 1.

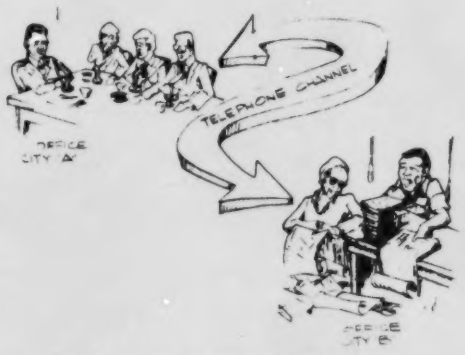

FIGURE 1. ROOM TO ROOM TELECONFERENCE

Room to Several Other Rooms. Mr. "A" of Sales Promotions Inc. next wishes to discuss the new promotion stunt with each of his regional offices, three in all. Since the comopany's transportation budget has been cut it is agreed that each of the regional teleconference rooms should be used. The five rooms are interconnected by communicatlions links as in Figure 2. Note, the addition of a bridge.

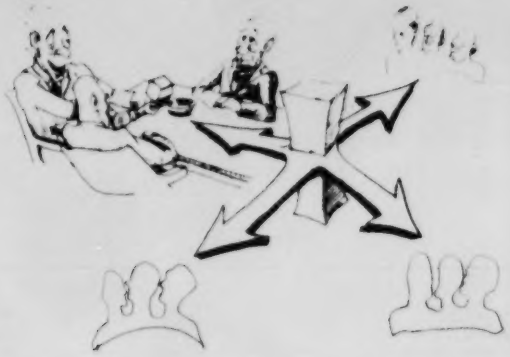

FIGURE 2. ROOM TO SEVERAL OTHER ROOMS TELECONFERENCE

Station to Several Other Stations. Having been appropriately motivated by Mr. "A" each of the regional managers are now faced with the responsibility of getting the new promotional package to each of the regional field offices. Station to station conferencing is used as pictured in Figure 3. Once again a suitable bridge is required as part of the communications link. More on this later.

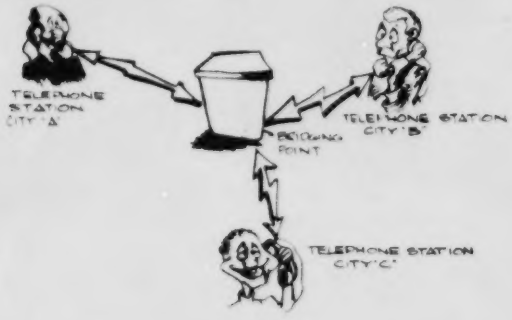

FIGURE 3. STATION TO STATION TELECONFERENCE

Various Other Combinations. On another occasion Mr. "A" and his staff were assembled in their conference room which was connected to the regional office at City B. Two members of this region could not be physically present so they were added as pictured in Figure 4. One member was reached at his home while the other, who was on the road, called in from a pay station in City C.

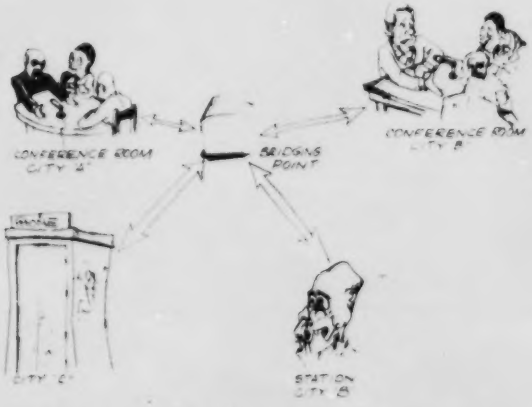

FIGURE 4. ROOM TO ROOM AND STATIONS TELECONFERENCE

Modes of Operation

Let us now focus our attention on another aspect of audio teleconferencing, the "modes of operation". Here an attempt is made to explain how the conferees depicted in the scenarios interact with each other. Ideally the audio teleconference system employed should allow complete vocal freedom (talker, listener activity) as in any normal verbal exchange. (Full duplex) Technical limitation may, on the other hand, limit this freedom allowing only one talker at a time (half duplex) and in still other cases the conference originator may choose to restrain interaction by "broadcasting" his thoughts or message. Consider the following.

Full Duplex. The audio teleconference between Mr. "A" and his staff at headquarters and regions were full two way exchanges. The teleconference system employed could have been configured as shown in Figure 5. But, as indicated in the figure, acoustic feedback problems and the need for bridging in some instances results in technical limitations which imposed a more disciplined verbal exchange.

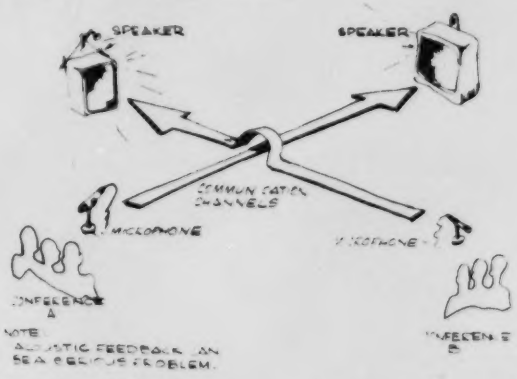

FIGURE 5. FULL DUPLEX TELECONFERENCE

Half Duplex. This more disciplined verbal exchange is depicted in Figure 6. The "talk, listen" switch may be operated either manually (push to talk) or automatically (voice actuated).

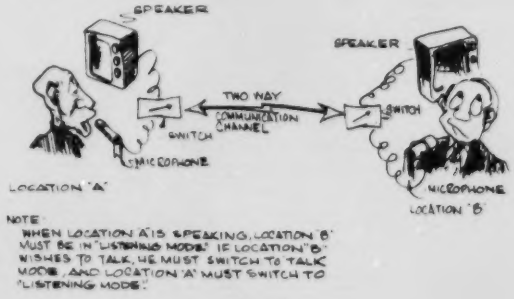
SPEAKER
SPEAKER
TWO WAY COMMUNICATION CHANNEL
SWITCH
SWITCH
MICROPHONE
MICROPHONE
LOCATION 'A'
LOCATION 'B'
NOTE
WHEN LOCATION 'A' IS SPEAKING, LOCATION 'B' MUST BE IN "LISTENING MODE" IF LOCATION 'B' WISHES TO TALK, HE MUST SWITCH TO TALK MODE, AND LOCATION 'A' MUST SWITCH TO "LISTENING MODE".

FIGURE 6. TWO-WAY HALF DUPLEX TELECONFERENCE

Broadcast. Quite frequently Mr. "A" of Sales Promotions Inc. chooses to make rather long presentations to his wide spread staff. On such occasions he does not wish to be disturbed by background noise at the outlying location. The broadcast mode pictured in Figure 7 satisfies these needs.

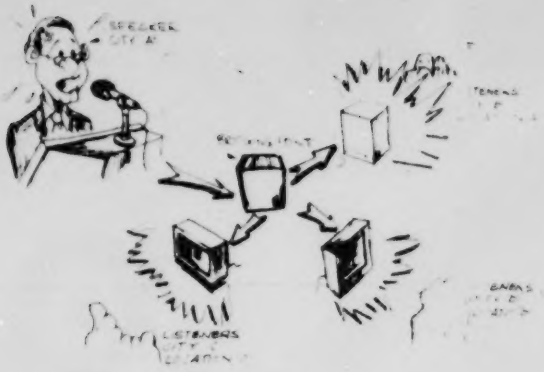

FIGURE 7. BROADCAST TELECONFERENCE

Methods of Establishing Audio Teleconferences

The various audio teleconferences pictured thus far may be established in any one of four ways. The conference rooms and/or stations may be hard wired as suggested by Figures 1 through 7 or they may have been established by a telephone operator of the Message Telephone System (MTS). Two other possibilities also exist. These are referred to as Ad-Lib and Meet Me arrangements.

Toll Conference Operator. Figure 8 depicts the conference originator at Telephone A being connected to conferees at B and C by means of operator controlled cord circuits associated with a suitable bridge located in the telephone company's central office. This system has been around a long time and in fact, is still the only system available on the MTS. Note, some of the newer private branch exchanges (PBX) are equipped so that other local stations may be added to an incoming call. Although these arrangements are often referred to as "conferencing" they are, in reality, only "add-on" features.

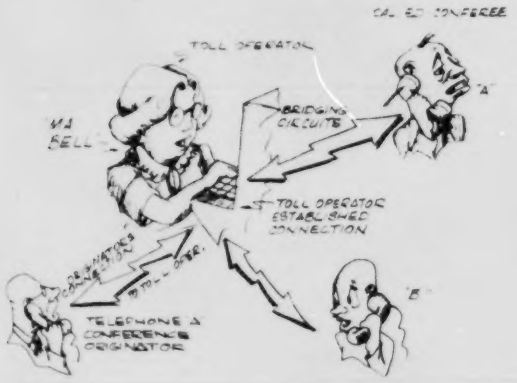

FIGURE 8. TOLL OPERATOR CONTROLLED TELECONFERENCE

Meet-Me. Normally when one wishes to call a meeting he first identifies a time and place generally acceptable to all or most proposed attendees. He than sends out a meeting notice which usually contains some sort of agenda and information regarding time and location of the meeting. Attendees hopefully show up at the designated place and hour. One could envision an audio teleconferencing system that would be managed in the same way. Instead of being advised of a meeting place the conferees would be given a telephone number to call at the proper hour. His call would be automatically answered and placed in the conference bridge along with all other conferees. This then is Meet-Me teleconferencing and may be pictured as in Figure 9.

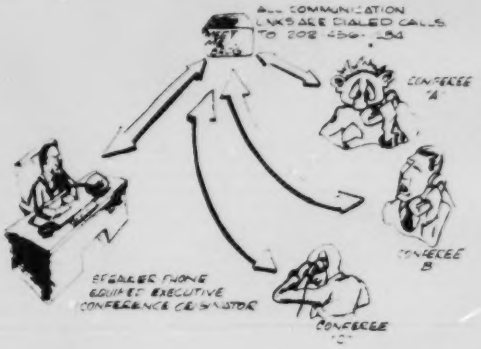
ALL COMMUNICATION LINKS ARE DIALED CALLS TO 202 456-1234
CONFEREE "A"
CONFEREE "B"
SPEAKER PHONE EQUIPPED EXECUTIVE CONFERENCE ORIGINATOR
CONFEREE "C"

FIGURE 9. "MEET-ME" TELECONFERENCE

Ad-Lib. Since many users of telephone conferencing services prefer the flexibility of managing their own teleconferences, arrangement must be provided that allows the conference originator the capability of dialed access and remote control of each audio teleconference. The remote control feature should give the originator the capability of dropping and redialing any conferee, selecting any conferee line for private discussion outside of the bridge, and blinding the received signal from any or all conferees. The blinding capability is useful to prevent conferee background noise from entering the bridge when the audio teleconference system is in the "Broadcast" mode. Audio teleconference systems equipped for remote control by the originator are referred to as Ad-Lib systems, see Figure 10.

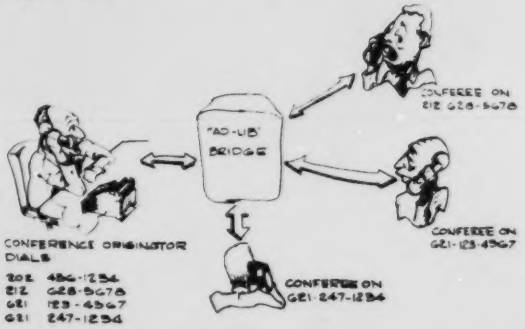

FIGURE 10. "AD-LIB" TELECONFERENCE

Interfaces

Thus far in this section of the paper we have been briefly reviewing audio teleconference arrangements, modes of operation and methods of establishing or assembling conferees by various interconnections of communications links. Detailed discussions of hardware arrangements have been purposely avoided since this paper is intended to concentrate on transmission systems and considerations. However, this section of the paper would not be complete without mentioning briefly how the audio sensing devices and reproducers; the microphones, loudspeakers and telephone handsets, are connected to the communications links. The "interface" or connection

between the audio device and the communications link will depend somewhat on the type of transmission system employed in the communications link and the audio instruments used at each terminal.

POTS. If the audio teleconference is established utilizing the plain old telephone (POTS) and the Message Telephone System (MTS) the interface is nothing more than the telephone set which is directly connected to the local loop and subsequently to the network through the local exchange. Since the MTS is basically a two-wire transmission system the interface device, the telephone set, will include a hybrid. The hybrid and its associated balancing network is employed to separate the transmit and receive signals as they enter and leave the local channel. See Figure 11. The hybrid is mentioned here since, as will be discussed later, its characteristics have considerable impact on the overall performance of the audio teleconferencing system. The MTS will be discussed in more detail subsequently.

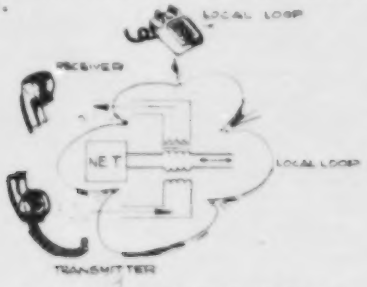

FIGURE 11 POTS INTERFACE THE HYBRID

It should come as no surprise that the POTS telephone set, slightly modified, could be arranged to interface with a private or dedicated communications channel which may be either a two-wire or four-wire transmission system. Two and four wire systems will be discussed subsequently. The telephone set hybrid would not be required in the fourwire system.

Speakerphone. The sensing and transducer elements of the telephone set may be replaced by the speakerphone which gives the teleconference originator or conferees "hands free" operation as was pictured in Figures 2 and 9. Of significance from a transmission standpoint is the fact that acoustic feedback now becomes a major concern. Review Figures 5 and 6. Although speakerphones can be modified to interface four-wire communications channels they are usually used on the two-wire MTS. In the speakerphone system the simple hybrid is replaced by a rather complex voice actuated switch which mutes the loud speaker when speech energy is present at the microphone. A typical speakerphone installation is pictured in Figure 12.

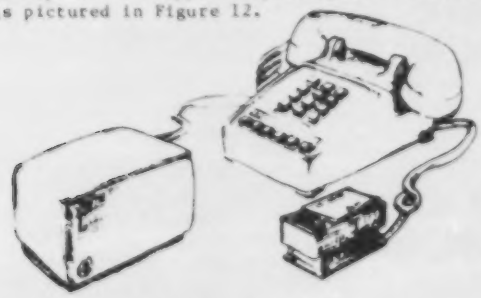

FIGURE 12 SPEAKERPHONE

The Bell Type 50A conference telephone is similar to the speakerphone but it is intended for portable use and may be readily connected to any local telephone loop. The 50A has two microphones, only one of which can be actuated at a time. As is the case of the speakerphone, room acoustics play an important part. In each case talkers should be within about a foot and a half of the microphone for best results.

Systems Requiring Sound Reinforcement. Because the speakerphone and 50A conference telephone are more or less restricted to use in small rooms, other arrangments must be made for large conference rooms and/or auditoriums when they are to be interfaced to communications channels. Since room acoustics play a significant role in the overall performance of such audio teleconferencing systems such rooms must generally be specially treated acoustically. This is a comprehensive subject in itself. Bell Telephone Laboratories has recently completed work covering the treatment of rooms to be used for audio teleconferencing. Although the work is yet unpublished the Bell System operating companies have been furnished suitable guidelines for their use.

Interfacing large conference rooms with communications channels requires a unit not unlike the heart of the speakerphone except that provisions are made for inputing several microphones and additional amplification is provided for delivering audio power to several loudspeakers usually located in the room's ceiling.

Microphone placement is especially critical. Bell Telephone Laboratory studies show that the allowable distance between talker and microphone bears a direct relationship to the room acoustics and generally must be within a few feet of the talker. Microphones suspended from room ceilings have been used in existing systems and probably will continue to be used wherever less than satisfactory audio teleconferencing can be tolerated. In cases where large auditoriums are to be interfaced with audio teleconferencing systems the existing audio sound reinforcing system must be interfaced to the communications link by means of a device similar to the ones described above. However, it must be pointed out that room acoustics that may seem satisfactory to occupants of the auditorium will very often be completely unsatisfactory to the listener on the far end of an audio teleconference network. Such systems have been made to work however but, obviously, it is not the intent to cover such systems here.

TRANSMISSION SYSTEMS

Types of systems; facilities; and bridging methods.

Types of Systems

What do we mean when we refer to a transmission system used in audio teleconferencing? A transmission system, as will be discussed in this section, is a collection of communications links interconnected by some form of bridging device so that two way exchange of information may take place between each end point of each communications link. Furthermore, each communications link is composed of one or more communications facility. In other words, the communications link extends between conferee interface and the bridge where as the facility is the physical media, wire or multiplex carrier, of which the link is composed. Transmission Systems may be divided into two major categories, identified here as either (1) dedicated networks or (2) the Message Telephone System (MTS).

Dedicated Networks. In keeping with the above definition of a transmission system Figure 13 shows several communication channels each extending from a bridge to a telephone set, speakerphone or treated conference room or auditorium. An additional communication channel is used to interconnect two bridges each of which may be located at points convenient to the telephone sets served. Each of the communication channels are composed of one or more facilities. The facility may be either two or four wire. Types of facilities will be addressed in later paragraphs. The actual electrical design of dedicated networks are complex and beyond the scope of this paper. However, some of the parameters affecting such designs i.e., loss, noise, circuit balance etc. will be addressed in the next section of the paper. It will be shown later that the number of stations that can be conferenced and the number of bridges employed depends on the distance between stations, which affects noise build up and the degree of electrical balance, echo control, that can be achieved in the design. Keep in mind that there will be some form of hybrid at each station. Each of these and the bridges, because they are not electrically perfect devices, will contribute some echo or feedback. In addition each facility may contribute some feedback. In fact there will be hybrids at the intersections of all four-wire and two-wire facilities. Obviously, all of the noise and echo contributed by each communications link will be added at the bridge and will be evident to each listener. Further discussions may be found in references.

FIGURE 13. DEDICATED TRANSMISSION SYSTEM

Message Telephone System (MTS). When maximum flexibility is required as was pictured in Figure 4, the Message Telephone System (MTS) should be considered. As shown in Figure 14, the MTS is composed of a complex network of switches arranged in tandem. Each station of the network is connected to its End or serving office by means of a two-wire local channel or facility. End offices, designated as Class 5, are connected to their serving Toll or Class 4 offices. Although the links between the Class 5 and 4 offices, the Toll Connect Trunks, may be made up of fourwire facilities, Class 5 and 4 offices are usually two-wire switches.

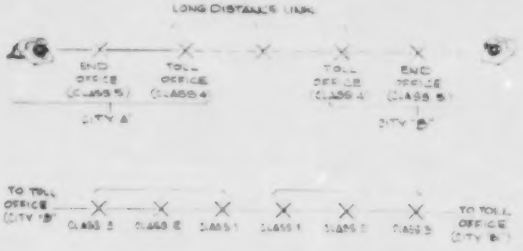

FIGURE 14. MESSAGE TELEPHONE SYSTEM

As pictured in the lower portion of Figure 14 Toll offices are interconnected by a network of trunks and higher ordered switches. These switches are designated as Class 3, 2 and 1 and are all four-wire switches interconnected by four-wire trunks. Not every Toll connection will be made up of links between all classes of offices. In some cases Toll offices may be connected directly to other Toll offices, or links passing through only one or two other intermediate switches. A detailed description of the MTS may be found in reference 1. The electrical characteristics of the MTS as related to audio teleconferencing will be addressed later.

Types of Facilities

Each communication link of an audio teleconferencing system, whether it be a dedicated network or one utilizing the MTS, will consist of one or more facility. These facilities may be either two or four wire cable or may be channels of a multiplex carrier system. Carrier systems are usually four-wire, that is the transmit and received directions are separated paths.

Two-Wire. Since the two-wire facility is composed of a single transmission path, bi-directional energy must be separated at each end of the facility by means of some form of hybrid so that the energy in each direction may be either amplified or, in the case of received energy, be used to actuate some form of transducer. See Figures 11 and 15. Energy on the transmit side of the hybrid is divided between the facility and the balancing network. In an ideal situation,

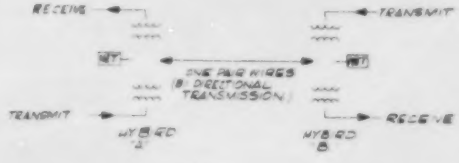

FIGURE 15. TWO-WIRE FACILITY

56

the energy divides equally and completely so that no energy is reflected towards the received side. Since, however, the network will only approximate the characteristics of the two-wire facility, the electrically perfect hybrid balance will never be achieved and some energy will always appear on the receive side. In dedicated systems precision balanced networks may be used to achieve a high degree of balance limiting reflected energy or echoes. However, in switched networks, such as in the MTS where facilities having different characteristics must be switched together, only a compromise balance can be achieved. These facts are of major importance to audio teleconferencing systems as will be shown in the next section of this paper.

Four-Wire. Four-wire facilities consist of two pairs of wire or two separated channels in the case of multiplex carrier systems. See Figure 16. Obviously, since there are no hybrid devices required to separate the two directions of transmission and since hybrid balance and echo are not factors the four-wire facility is ideally suited for audio teleconference systems.

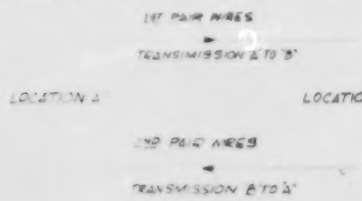

FIGURE 16. FOUR-WIRE FACILITY

Bridging Methods

Our purpose here is only to relate networks and facilities to audio teleconferencing systems pointing out items of significance to be considered in the design of such systems. Transmission requirements that have been established for two point telephone services generally apply to audio teleconferencing systems. These requirements are described in various engineering documents and accordingly need not be repeated.

Interconnecting three or more conference locations requires use of bridging hardware to interconnect the transmission path to each of the locations. Ideally, the conference bridge should cancel echos generated internally to the bridge, protect conferees from external noise and echo and have minimal transmission loss. Two methods of bridging will normally be encountered in audio teleconferencing systems. These are referred to here as Mid-Point and as End-Point bridging.

End-Point Bridging. Figure 17 is intended to portray the situation where the bridging element of the audio teleconferencing system is located at one end of the network. In this arrangement one station, usually the originating station, is connected to several remote stations via independent communication links. Assuming the approximate distance between the originator and each conferee (Points A thru D of the figure) is about equal, it is evident that the electrical or transmission loss is about twice as much between each conferee as it is between conferee and originator. This is referred to as loss contrast and can be a serious problem in audio teleconferencing systems. Ideally, this loss contrast should be minimized by placing the bridging element midway between each conferee and originator. There are other reasons, however, as will be discussed, for Mid-Point bridging.

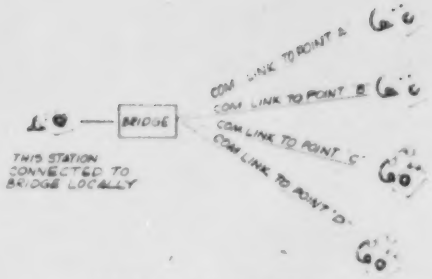
BRIDGE
COM LINK TO POINT A
COM LINK TO POINT B
COM LINK TO POINT C
COM LINK TO POINT D
THIS STATION CONNECTED TO BRIDGE LOCALLY

FIGURE 17. END POINT BRIDGING

Mid-Point Bridging. Mid-Point bridging, as shown in Figure 18, is relatively easy to accomplish in the design of a dedicated audio teleconferencing system. Such dedicated systems can be designed so that the electrical loss is nearly equal between all points on the network. This is even true if more than one bridge is used if the link between bridges is designed to have zero loss or unity gain. See Figure 19. Audio teleconferencing systems utilizing the MTS, of necessity, must utilize End-Point bridging since the center of the MTS is not readily accessible. However, considerable work has been done developing an audio teleconferencing system which utilizes Mid-Point bridging on the MTS. This system utilizes bridges placed within the switched network at the Class 3, 2 or 1 switching points. Review Figure 14. The desirability of Mid-Point bridging, especially in switched network environments, will become evident in the next section.

FIGURE 18. MID POINT BRIDGING

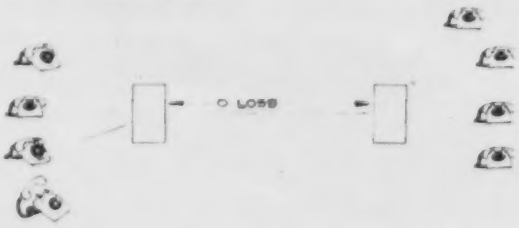

FIGURE 19. MID POINT BRIDGING TWO BRIDGES

TRANSMISSION CONSIDERATIONS

A discussion of some parameters effecting transmission; impact of the parameters on Audio Teleconferencing Systems; noise and echo treatment; and bridging elements.

Some Parameters Affecting Transmission

Of the many electrical parameters affecting transmission on any two-point voice band communication system, see reference, the three that are of prime consideration to designers of audio teleconferencing systems are loss, noise and echo. These three are to be addressed here.

Loss. Connections established over the switched network will typically have an end to end loss of about 14 dB. Direct connections (private line or dedicated networks) are designed to have an end to end loss of 10 dB or 16 dB respectfully for two-wire and four-wire services. Since, as stated above, transmission requirements for two point switched and non-switched or dedicated services apply to audio teleconferencing systems the following discussion will be developed by considering a typical two point connection established over the MTS. The Bell Laboratories 1969 - 1970 connection survey[2] showed that the 1,000 Hz loss between End offices could vary by several dB but the mean measurement of all test connections was 6.7 ± 6 dB with a standard deviation of 2.1. Loop surveys have shown that, as pictured in Figure 20, the end to end loss for two point connections will generally approach 14 dB. As indicated, trunks between switches are engineered in accordance with Via Net Loss (VNL) design. Simply stated, VNL design takes into account the link and types of facilities and their delay characteristics so that the overall loss of typical built up connections will be sufficient to compensate for echo delay. Typically, trunk losses between switches will be less than 1 dB. Note that at least 5 dB of loss between End offices is assured by the 2.5 dB fixed loss designed into each Toll connect trunk.

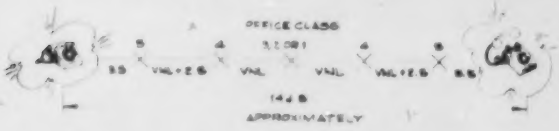

FIGURE 20. LOSS (TWO POINT CONNECTION)

Since all communications links are subject to echo due to impedance imperfections and since a talker's tolerance to echo can be controlled to some extent by introducing loss into the echo path it follows that some end to end loss must be built into all connections. As it turns out, approximately 14 dB seems to be a reasonable figure around which to build our discussion of noise and echo.

Noise. Noise as it relates to communications links is discussed in a Bell System technical reference[3] and is repeated here in part.

Noise on telecommunications channels arises from numerous sources. Some of these are thermal noise from amplifiers, 60 Hz and its harmonics which may be picked up from power line induction, intelligible or unintelligible cross talk, single frequency tones, and switching and signaling transients. All of these effects and others come under the general heading of noise.

Background noise has been referred to in various places as white noise, Gaussian noise, message circuit noise, etc. What is intended is a frequency weighted measure of the total power on a channel not arising from the desired signal. The frequency weighting used is called "C-Message" weighting. See Figures 21 and 22.

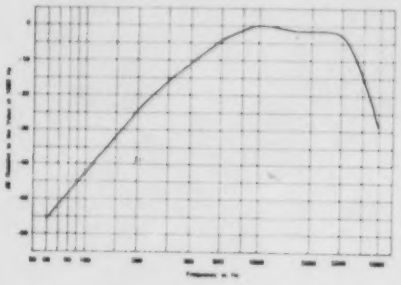

FIGURE 21. C-MESSAGE WEIGHTING CHARACTERISTIC

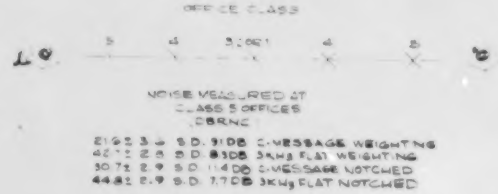

FIGURE 22. NOISE TWO POINT CONNECTION

Echo. As stated in the Bell System technical reference[4] on transmission parameters, impedance mismatches in a circuit give rise to reflections of signal energy called echoes. A single reflection causes energy to return to the transmitter (talker echo). If the single reflection is again reflected at an impedance mismatch, signal energy will arrive at the receiver sometime after the original signal (listener echo). Echoes due to multiple reflections also occur, but they are usually insignificant by comparison with those due to single or double reflections.

As stated earlier echoes generally cause no problems on four-wire channels. However, on two-wire channels, and on channels containing both two-wire and four-wire facilities, echoes may be interfering to voice combinations. Echoes are controlled by matching impedances and by controlling loss in echo paths.

As was shown in Figure 15 hybrid transformers are used to join two and four-wire facilities. The impedance of the balancing network ideally is the same as the impedance looking into the two-wire facility. If this condition is met, signal power from the received side of the four-wire facility will divide equally between the balancing network and the two-wire facility and no signal power will "return" on the transmit side of the four-wire facility. In practice, the match is never perfect. Morever, the match is different at different frequencies. Some portion of the received signal power is thus returned on the transmit side of the four-wire facility.

Various tests are used to determine channel quality with respect to echoes. These include return loss, echo return loss, singing return loss, singing point and singing margin.

Return loss is the ratio in dB of the power of a single frequency signal placed on the received side of the four-wire facility (Figure 15) to the resulting power at the frequency appearing on the transmit side.

Echo return loss is a weighted average of the power returned to all frequencies between 500 and 2,500 Hz and then expressed in dB.

Singing return loss is also a weighted average of the reflected power at all frequencies in a frequency band expressed in dB. There is a low frequency test covering the 200 to 500 Hz band, and a high frequency test covering 2,500 to 3,200 Hz band.

Methods for performing these various tests are discussed in the Bell System technical reference.

Impact of Transmission Parameters on Audio Teleconferencing Systems

Thus far we have reviewed loss, noise and echo as related to typical two point connections on the MTS. Had we directed our attention to dedicated networks we would have seen that measurements of these parameters would have revealed similar figures. Generally, these parameters are more easily controlled on dedicated communications links. Accordingly, dedicated systems may be engineered to have specific end to end losses, and noise and echo requirements held to closer tolerances. Since audio teleconferencing systems are made up of several two point connections bridged at an appropriate point or points we will now turn our attention to the overall effects of loss, noise and echo when combined at the bridging point. This will be done by a further examination of echoes followed by more on noise, loss and control.

Combined Echoes. Since we are primarily concerned with talker echo let us turn our attention to speech signal levels (measured in VUs) as they may appear on a typical two point connection. See Figure 23. Although a typical talker's speech level will vary widely at the interface of his telephone instrument and the local channel, a figure of -13VU could be assumed Assuming in addition, an end to end loss of 14 dB, the speech energy would reach the listener at approximately -27VU. The distribution of speech energy along the circuit might typically be as shown on the figure.

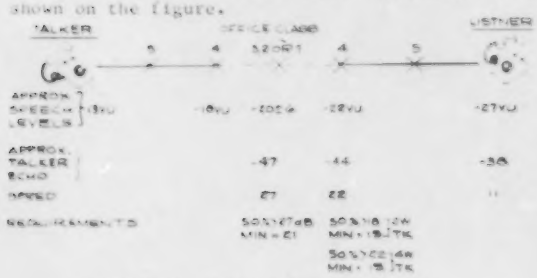

FIGURE 23 ECHO TWO POINT CONNECTION

Now let us consider the resulting talker echo, that is, the amount of reflected energy at various points along the circuit. If all design objectives have been met we could expect to measure reflected energy as shown on the figure.

Specifically, as shown, 50% of all such measurements should exhibit reflected energy levels greater than 27 dB below the signal energy at the mid-point of the circuit. In other words, since the signal energy at the mid-point measures minus 20 VU, the reflected energy at that same point must measure greater than 47 VU on at least 50% of all connections. The minimum acceptable level would be 41 VU or reflected energy 21 dB below the signal.

It is important to note that the spread between signal levels and the level of reflected energy varies at different points along a typical circuit. If all design requirements are met this spread varies from 27 dB at the circuit mid-point to 11 dB at the interface between the telephone instrument and its local channel. The reason for this may become clearer when one realizes that the reflected energy is reduced by the amount of loss in the circuit as the reflections are (propagated) back towards the talker. In addition, keep in mind the fact that, since the center links are all four-wire, most of the reflected energy is developed in the two-wire portions of the circuit and will be reduced by the amount of loss in the four-wire sections.

Let us now assume that we are going to place a bridging device at some point in the two point connection. The bridge is intended to distribute the talkers energy, ideally without loss, to each of several other connections. See Figure 24. Likewise, reflected energy from each of the other connections will be combined at the bridge and returned to the talker as shown on the figure. The amount of energy returned to the talker will be the power summation of all reflected energy received at the bridge. Thus if one considers each bridge port to be a hybrid reflecting energy the amount of reflected energy becomes greater as more and more legs are added. The spread between the signal energy and the reflected energy before the addition of legs may be thought of as the initial hybrid balance which is reduced by 3 dB every time the number of legs added to the bridge is doubled.

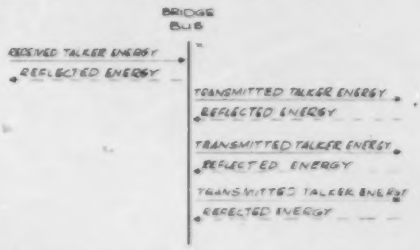

FIGURE 24. DISTRIBUTION OF TALKER ENERGY AT BRIDGING POINT

Specifically, let's assume a bridge is placed at the interface point between the telephone instrument and its local channel. (End-Point bridging). The reflected energy at this point is approximately 9 dB below the signal therefore our assumed hybrid balance is 9 dB. Now as we proceed to add legs to the bridge this balance will be reduced by 3 dB everytime the number of legs is doubled. In other words, adding the first leg reduces the balance to 6 dB, adding two more legs reduces it to three dB and adding four more legs further reduces the balance to 0 or the point at which the circuit becomes instable. See Figure 25.

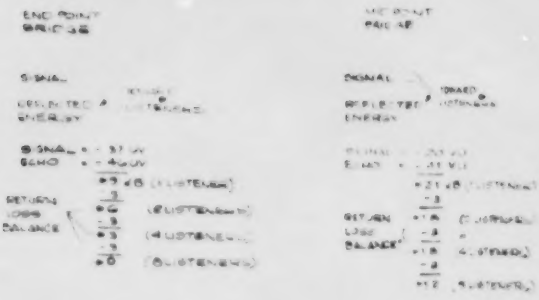

FIGURE 25 DISTRIBUTION OF ECHO AT BRIDGING POINT

Now consider what happens if the bridge is moved to the center of an audio teleconference connection. (Mid-Point bridging). Since the reflected energy at this point is 21 dB below the signal, an assumed hybrid balance is 21 dB. Adding the first leg reduces the balance by 3 dB to 18 dB. Adding two more legs reduces it to 15 dB and finally the addition of four more legs reduces the balance to only 12 dB. The significant point to grasp here is that the bridge, when located at the center of the network, still has a 12 dB margin for circuit stability after adding 8 legs to the bridge as opposed to 0 dB after adding 8 legs to an end-point bridge. See Figure 25.

In the above discussion it was assumed that the bridge element did not contribute any reflected energy. In other words it was assumed that the bridge was a perfect device and the separation between talker energy and reflected energy was infinite. Such is not the case. At best, a perfectly terminated resistive bridge will reflect talker energy at levels only some 40 dB lower. More will be said about this later.

Combining Noise. Noise appearing on each leg is added to the bridge element as in any circuit exposed to several noise sources. Obviously, the bridge should be placed in a circuit or network at the point where each leg contributes the least amount of noise. Referring again to Figure 22 one would assume that mid-point bridging is preferred.

Noise and Echo Treatment

Reviewing the above discussion it is evident that within an audio teleconferencing system some provisions are needed for controlling echo and noise. Quantitatively speaking the point at which such treatment becomes necessary depends upon the number of conferees to be interconnected and the location of the bridge within the network. Designers of audio teleconferencing systems are then faced with two questions, when and how. When is treatment for noise and/or echo necessary and how is the treatment to be accomplished? This section deals with these questions but will not go into detailed hardware specifications.

Treatment For Noise. In considering noise treatment two situations should be addressed. Situation 1 consists of a single communications link between two conference rooms equipped for audio teleconferencing. Situation 2 consists of individual communications links between several audio teleconference rooms or stations. In the first case special treatment will generally not be required because the noise contributed by the single link should be within tolerable limits. In the second case, however, each communications link is a noise contributor and although each link could be within tolerance the combined noise of several links could result in an unacceptable noise level to each listener. Since, in a well disciplined conference situation, only one bridge port should be active (talker energy present at bridge) at any instant, the obvious solution to control noise is to terminate all inactive ports. The port terminations could then be lifted automatically by the incoming speech energy whenever a port becomes active.

Switched gain amplifiers are devices that contain special circuits for detecting the presence of speech energy. Whenever speech energy is present pads of some predetermined loss are switched out restoring the activated port to full sensitivity. Therefore, a switched gain amplifier containing 15 dB pads would reduce noise levels at the input to a bridge port by 15 dB whenever the port is idle. In other words 25 dBrnc of noise at the input to a switched gain device would appear at the bridge at a level of 10 dBrnc.

Treatment for Echo. From the above discussion one can see that pads in each port leg would protect against echo build up as well as noise. It should also be noted that a pad in the transmit leg, which would normally be switched "out", could be switched "in" whenever energy is present on the received side of that port. This action would reduce the level of echoes returned to the talker that may arise within the bridging element.

Treatment Using Switched Gain. A typical voice switched gain unit consists of two variable gain amplifiers, one in the transmit path (input to the bridge) and the other in the receive path, and a differential control circuit. When used with a two-wire connecting unit the gain of each amplifier can be varied electronically from 0 to 15 dB, depending upon the speech levels received at the bridge port. The amplifier in the transmit path normally has no gain, so that the constant 15 dB attenuates weak input signals (such as noise) before they enter the bridge. The energy of a conventional voice causes an increase in gain of the amplifier effectively canceling out the 15 dB constant loss; and the speaker's voice enters the bridge without attenuation. This feature prevents the noise present on each bridge connection from building up to unacceptable levels when many connections are made to the bridge.

Switching of gain should not be detectable to the conferee listeners. The three most important parameters involved in the switching action are the amount of switch gain, time constants and threshold slope.

As previously mentioned, the loss required for adequate echo control and bridge stability is 15 dB. The time constants involved in switching gain are attack time and hangover time. These two values were selected. by a compromise. The attack time has to be short enough not to clip the first syllable of speech and long enough not to be sensitive to impulse noise. A range of 1 to 5 milliseconds has been judged best. The hangover time must be long enough to prevent false operation or echoes and to prevent noticeable gain changes between syllables; if it is too long, however, a rapid interchange between

talker's will be hampered (i.e. the amplifiers will not switch fast enough). The hangover time best meeting these requirements has been found to be between 100 and 150 milliseconds. As a result, the design requirements are 3 plus or minus 2 milliseconds for attack time and 150 plus or minus 75 milliseconds for hangover time.

A "soft-switch" is required in order to prevent sudden drops in level during a weak talker's speech since some syllables will be too weak to keep the switch on. Soft switching renews the loss gradually as the talker's voice goes through the threshold, thus considerably reducing the sudden change in level affect. The threshold level (point where a talker's speech volume completely switches out the loss) is, therefore, a slow instead of a single cutoff value. Figure 26 shows the attack and hangover time constant for various talkers' level and Figure 27 illustrates the relative switching aspects of soft vs. hard switching.

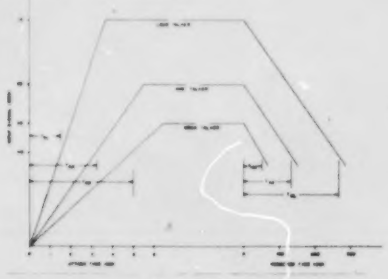

FIGURE 26. ATTACK AND HANGOVER TIMES OF SWITCHED - GAIN AMPLIFIERS

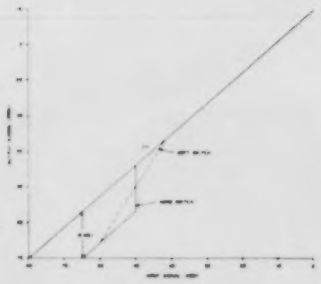

FIGURE 27. SWITCHED - GAIN PARAMETERS

Other design considerations for switched gain amplifiers include the dynamic range of talkers and the maximum power requirements. These two factors determine the threshold level. It has been found that the loudest speech rarely exceeds +7 dBm and the weakest speech seldoms drops below -40 dBm; therefore, the threshold sensitivity which will completely switch the loss from the transmit to the receive side is set at -40 dBm. A switching guard circuit prevents a small amount of speech energy, returning as an echo, from operating the amplifier in the transmit path and a noise operated gain adjusting device prevents noise as high as 55 dBrnc from operating the switch.

Typical Switched Gain Amplifier. A typical switched gain amplifying device is shown (schematically in Figure 28). The switched gain amplifier unit consists basically of a receive amplifier, a transmit amplifier and electronically controlled pads which controls the gain through the receive and transmit legs of each bridge port.

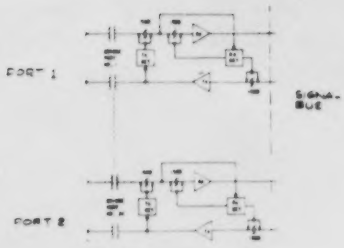

FIGURE 28. TYPICAL SWITCHED - GAIN AMPLIFIERS

The receive leg has two pads, an Echo Pad and Signal Pad. Each pad introduces a 15 dB loss. The Signal Pad is "IN" in the idle condition and switches "OUT" when an incoming voice signal is present at the receive amplifier input. The Echo Pad is "OUT" in the idle condition and is switched "IN" whenever a signal is present at the transmit amplifier output.

The transmit leg has one pad which has a loss of 15 dB (24 dB option). This pad is "OUT" in the idle condition and is switched "IN" whenever an incoming signal is present at the receive amplifier input.

The Signal Pad in the receive and transmit leg are controlled by circuitry consisting of an amplifier, a limiter, a voice filter, and a peak detector. The DC signal controlling the pad is monitored by a light-emitting diode (LED) Lamp on the front panel of each switched-gained amplifier module. This lamp, the RA Lamp is "ON" when a voice signal at a receive amplifier exceeds a certain minimal level (adjustable by the receive amplifier sensitivity potentiometer).

The Echo Pad controlled circuitry consist of an amplifier and peak detector. The DC signal control of the Echo Pad is monitored by a LED Lamp, the TA Lamp, on the module panel.

Thus, a voice signal at 0 dBm level, incoming to the input to Port 1 reaches the output of the same port attenuated over 60 dB (return loss factor). The same signal also reaches all other ports at 0 dBm or some predetermined level. In other words, there is a large loss from the input of the port to its output, but there is no loss or some predetermined small loss or gain to all other outputs.

Bridging Elements

The bridging network has one primary function. It must connect all conferees together so that transmission between any two ports have low loss and the echo returning to the talker is very low. (The application of bridges in certain system designs may include gain devices to offset losses inherent in the system external to the bridge.)

In this discussion bridges are classified as either passive or active. Passive bridges, more commonly referred to as resistive bridges, have been used almost exclusively in earlier multipoint private networks and audio teleconferencing systems. Active bridges, bridges containing active elements, have come into use more recently concurrent with the developement of transistors and integrated amplifier devices. Little will be

said about resistive bridges. Two types of active bridges will be discussed.

Resistive Bridges. Resistive bridges may be described as resistive hybrids in each port. Individual balance adjustments (impedance matching on each port) and the use of metallic film resisters for each port allow the stability and echo return loss requirements to be met but not easily when compared with the advantages of active bridges. A perfectly balanced resistive bridge will restrict the return loss (RL) to about 40 dB at best. As will be seen, this figure can be improved substantially using an active bridging element.

Active Bridges. Active bridges have been employed in audio teleconferencing systems in recent years more or less experimentally. With the exception of the Time Division (TD) bridge (to be described) active bridges are not generally available "off of the shelf" but can be fabricated utilizing solid state integrated circuits such as operational amplifiers.

TD Bridge. The TD conference bridge package currently available provides 24 port bridging capability and is designed to serve large conferencing requirements. The TD conference bridge compares favorably with a perfectly balanced four-wire resisted bridge and has the added advantage that it cannot be unbalanced by shorted, opened, or missmatched ports. Theoretically, TD bridges should have an infinite return loss, due to stray capacity and coupling between wires, however,some finite return loss is to be expected.

The bridge is designed for central office installation and utilizes a 10 and 1/2 inch by 23 inch equipment cage. The bridge is designed for use with a negative 24 volt filtered DC supply.

The bridge is composed of several plug in modules. A complete bridge package includes 24 switched gain amplifiers of the type described above, 13 filter switching units, a scanner unit, and a maintenance field test panel. Each of 12 filtered switching units accommodate two ports. The 13th filtered switching unit is used as part of the maintenance panel. Two or more TD conference bridges may be used intandem or a bridge can be split in various ways to function as two or more bridges of less capability.

Each bridge port includes a switched gain amplifier and a filter switching unit. Filter switching units are controlled by pulses from a scanner unit and are interconnected by a common bus wire as pictured schematically in Figure 29. The switched gain amplifier unit consist bascially of a receive amplifier, a transmit amplifier and electronically controlled pads which control the gain through the receive and transmit legs of each port. The operation of these amplifers are as described above.

Each filter switching unit contains two identical circuits, each serving one bridge port. Voice signals enter the input low pass filter and are "keyed" by an electronic switch to the central or bridging bus wire. Signals on the bus are then fed through a second electronic switch and low pass filter in the transmit legs of each port. The electronic switches are driven by control pulses from the scanner in such a way that each receive leg is conducting only one 24th of each scanning cycle. Transmit legs are conductive twenty three 24th's of each scanning cycle. Thus, each receive leg is turned "ON" (connected to the bridge bus) for only about 3 microseconds during each scanning cycle. During this "ON" period, the corresponding transmit leg of the port is turned "OFF" (disconnected from the bridge port). In other words, the voice signal reaches the bridge bus wire but is blocked and cannot enter its own transmit leg. This prevents feedback and results in high return loss. The signal, however, can reach all other transmit legs of the bridge.

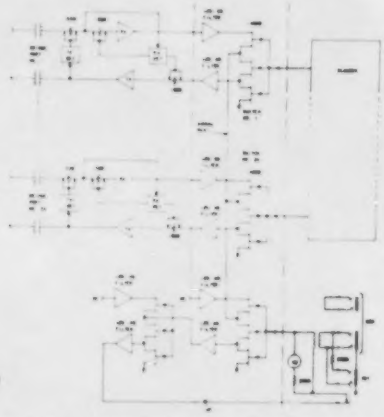

FIGURE 29. TRANSMISSION PATH TIME DIVISION BRIDGE

The scanner unit consist of a 300K Hz multivibrator, a binary counter, two "1 of 16" decoders, and a bi-stable flip-flop circuit. The first "1 of 16" intergraded decoders decodes clock pulses from 1 to 12; then the flip-flop switches the clock pulses to the second decoder which counts from 13 to 24. The 24 outputs of the scanner are sequential pulses at the rate of 12.5K Hz. Each output keys an appropriate switching unit. The scanner is powered by a 5 volt DC Zenner regulated source derived from the 24 volt central office supply.

The maintenance panel consists of jacks for each port and a VU meter for use in lining up the switched gain amplifiers. The VU meter is connected to the bridge bus by means of a filter switching circuit. It should be noted, however, that this filter switching unit is not keyed but is turned "ON" at all times.

Active Bridge Utilizes Selective Summing. A second type of active bridge receiving increase attention currently may be described as one consisting of operational amplifiers tied together using selective summing techniques. A bridge of this type is pictured in Figure 30.

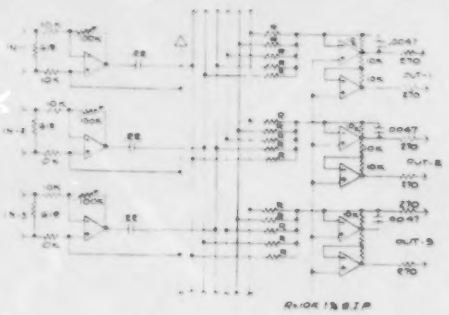

FIGURE 30. ACTIVE BRIDGE UTILIZING SELECTING SUMMING

Discussions of selective summing techniques may be found in currently available literature. The major advantage of these bridges are that they have extremely high return loss factors. In addition such bridges can be easily cascaded and are cost effective.

Selective summing is achieved by the "R" resisters as shown in the figure. The addition of another operational amplifier to develop phase inversion is shown in Figure 31. The addition of phase inverson to the selective summing bus improves the return loss factor still more. Figures of 80 dB or more are achievable.

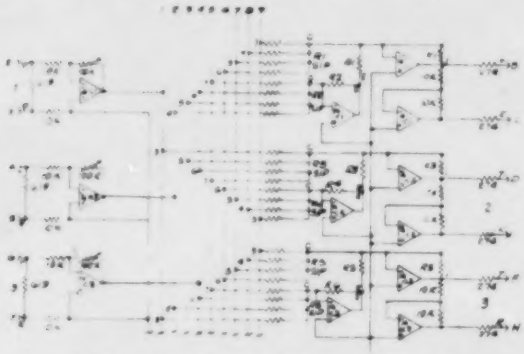

FIGURE 31 ACTIVE BRIDGE UTILIZING SELECTIVE SUMMING & PHASE INVERSION

CONCLUSIONS

Some recommendations and
a glimpse at the future.

Recommendations

During the past several years increasingly large numbers of audio teleconferencing systems have been coming into existence. Many of these systems have been provided to meet the needs of specific customers and have employed dedicated networks. Most of these systems have been developed without the benefit of suitable industry standards. Much impressive work has been completed by designers of hardware both within and external to the Bell System. Most of the techniques system designers employed stem from work at Bell Laboratories dating back to the earliest days of private line and toll conference services. Still more recent work within the Bell companies and at Bell Telephone Laboratories on audio teleconferencing systems utilizing switched private networks and the MTS suggests several recommendations that are presented here for consideration. These recommendations include bridge placements, terminal units and conference rooms to be equipped for audio teleconferencing.

End-Point vs. Mid-Point Bridging. Discussions regarding noise and echo in the preceding section of this paper substantiates our recommendation that whenever audio teleconferencing systems employ switched networks, either private or MTS, bridges should be placed in the electrical center of the network. Such placement reduces loss contrast between stations and improves circuit stability, i.e. singing margin. Refer to Figures 18, 19, 23, 24, and 25.

It is realized that certain customer requirements can only be met economically by employing end point bridging. These situations usually occur in dedicated private networks. In such situations the use of active bridging employing selective summing and phase inversion techniques is recommended.

Regarding Use of Speaker Phones. The use of speaker phones or similar devices on audio teleconferencing systems should be limited especially on systems interconnecting large numbers of conferees. The reasoning for this is as follows: Audio teleconferencing systems interconnecting large numbers of conferees most likely will employ switched gain amplifiers on each bridge port. The combined effects of voice switching at the speaker phone and at the bridge port often results in severe speech clipping. Experience has shown however, that these undesirable effects are reduced substantially when the Bell Type 4 Speaker Phone is employed at each station. In addition, whenever speaker phones are employed for voice teleconferencing systems users should keep in mind the fact that speaking within a reasonable distance of the speaker phone unit is essential.

Regarding Rooms Requiring Sound Reinforcemnt. Detailed recommendations regarding the interconnection of conference rooms requiring sound reinforcement (very large rooms or auditoriums) are beyond the scope of this presentation, however, a few general comments are appropriate. As mentioned earlier in the paper, Bell Telephone Laboratories has recently completed extensive work regarding the acoustic treatment of such rooms and have produced a set of guidelines dealing with microphone and speaker placement as well as acoustic treatment. Since this paper is yet unpublished it is not available for general distribution.

Our recommendations are as follows: Rooms other than auditoriums should be interfaced with communications links by utilizing specially designed hardware packages containing voice switching devices and sound reinforcing amplifiers equipped to handle up to four microphones and speakers. Although the speakers may be mounted in the room ceilings, microphones generally must be placed within 1 1/2 to 3 feet of the talker. The rooms, in most cases, will require acoustic treatment.

Large auditoriums will generally require specially engineered interfaces between the communications link and the sound reinforcing equipment normally employed within the auditorium. Two way full duplex conferencing in such cases is extremely difficult. Since proper microphone placement is of prime importance it appears that each situation must be treated independently on a "per case" basis.

A Glimpse at the Future

To be complete any discussion on audio teleconferencing must contain some thoughts regarding the future. The views expressed in the final paragraphs of this paper are the authors and do not necessarily reflect the views of his company's associates or company's policies regarding audio teleconferencing. The author's views are presented here to stimulate more thoughts and discussions concerning future audio teleconferencing. The author's views are presented here to stimulate more thoughts and discussions concerning future audio teleconferencing needs with the hope that such actions will eventually lead to the market research and technical development work necessary to turn proposals and plans into realities.

Direct Dial Conferencing (DDC). A trial is currently underway to evaluate a new Bell System service that will provide for the direct dial and control of an audio teleconference from any Touch Tone equipped telephone station. This service will give a conference originator the capability of directly dialing up to any six conferees reachable on the MTS. (The trial restricts bridge access to originators within the Minneapolis, St. Paul calling area.) This new service provides automatic billing. In addition it allows the originator to drop and redial any conferee or to remove a selected conferee from the bridge for private discussion. Although the system is currently limited to conferences of up to seven inviduals it is conceivable that future systems will be developed to handle larger conferences on a dialed up basis.

Meet-Me Systems of the Future. The DDC trial discussed in the preceding paragraph is the Ad-Lib type pictured in Figure 10. Future conference systems utilizing private switch networks or the MTS will most likely also provide Meet-Me capabilities such as was pictured in Figure 9. Educational seminars and "Monday morning" staff meetings could be handled very efficiently on a Meet-Me basis. Such systems

have a further advantage in that each conferee would be automatically billed separately as though he had place a conventional toll call. The originator would only be billed for the originating connection to the bridge and possible for bridge usage.

Possible unauthorized entry into the Meet-Me bridge has been viewed with considerable concern. Future Meet-Me systems would be equipped with the necessary logic to give the originator the capability of "unlocking" only those ports required.

Graphics. Users of audio teleconferencing systems are looking to the communications industry to provide the capability of simultaneous voice and graphic transmission over a "dialed-up" connection. This capability could be provided by either establishing a second dialed-up connection between graphics machines or by using some form of interdispersed voice and data transmission over a common teleconferencing system.

Manufacturers of graphics terminals should give consideration to the "impact" of "mid-point" bridging on their system design. Future graphics hardware should include polling capability to achieve syncronization between the transmitter at the originator point and receivers at remote points. Another approach would be to provide syncronization techniques that do not require "hand-shake" routines.

Conferee Interrupt. In those situations where the audio teleconferencing originator chooses to use the "broadcast mode", see Figure 7, the teleconference system should include a response system that would give the conferee listener the capability of "raising his hand" to signal his desire to speak. This action should be carried out so as not to disturb all other listeners. Recalling that, in the "broadcast mode", all conferee transmit paths to the bridge are normally blocked, one could imagine a system wherein the conferees transmission path to the bridge could be used as a signal path to be connected to logic at the bridge capable of storing "conferee signals" and retransmitting these over the originator's receive path to a suitable device provided at the originators lectern. Thus, the use of this "student to teacher" link would not interfere with the lecture since the signal paths would by pass the bridging element.

Computer Teleconferencing. Computer Teleconferencing is not new. It has been around for some time now in the form of computer time share systems. Such systems, when properly programmed could be used for computer to computer conferencing. However, one need not stretch his imagination too far to envision an audio teleconferencing system fully controlled by mini-processors that would establish preprogrammed conferences at appropriate times automatically dialing all intended conferees. Such a system might even recall an absent conferee at a later time to "replay" the conference.

The Ultimate Audio Teleconferencing System. Finally in conclusion and by way of review, let us look at the capabilities of ultimate audio teleconferencing system. Such system would:

1. Be available to any telephone station on the MTS network
2. Utilize the MTS network for maximum flexibility
3. Utilize mid-point bridging to maximize transmission performance
4. Utilize latest solid state active bridging elements to optimise transmission.
5. Provide for complete originator control
6. Handle small (up to 6 conferees) or large (up to some practical limit) conferences
7. Provide Meet-Me as well as Ad-Lib capabilities.
8. Provide for a broadcast mode.
9. Provide for conferee signalling during a "broadcast". Appropriate "Add On" terminal equipment would be available to the originator on short notice for "one short" needs
10. Provide equipment to interface to appropriately equipped conference rooms and auditoriums. Such equipment would also be made available as portable units for one shot requirements
11. Utilize latest solid state active bridging elements to optimize transmission
12. Allow originator capability of selecting bridging location to minimize his cost. Selecting bridges would improve network efficiency by reducing trunking needs
13. Provide for the transmission of graphics simultaneously with speech.

Since complete communications and understanding between individuals involves more than just the spoken word the ultimate would be a visual as well as an audio teleconferencing system.

References

1. Notes on Distance Dialing
American Telephone and Telegraph Company

2. Bell System Technical Reference (Pub 41007)
1969-70 Switched Telecommunications Network
Connection Survey (Reprints of Bell System
Technical Journal Articles - April 1971)

3. Bell System Technical Reference (Pub 41500)
Data Communications Using the Switched
Telecommnications Network - May 1971

4. Bell System Technical Reference (Pub 41008)
Transmission Parameters Affecting Voiceband
Data Transmission - Description of Parameters
July 1974

5. Operational Amplifiers Design & Applications
Burr-Brown Research Corp. McGraw Hill Book
Co. Page 429

TELECONFERENCING STATION EQUIPMENT

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Introduction

The purpose of this paper is to review the requirements for telephone conferencing station equipment used on 4-wire private line and voice-grade dialed telephone circuits. Station equipment will be confined to that intended for groups of individuals at multiple locations. Further, it will be shown how present Darome equipment satisfies these requirements. Ideas on what improvements can be incorporated to improve the technical aspects of conferencing will be presented.

4-Wire Private Line Station Equipment

After seven years of experience operating Wisconsin's Educational Telephone Network, the Instructional Communications Systems personnel in conjunction with Wisconsin Bell defined the requirements for multipoint, private line station equipment. These requirements defined a product to be used in classroom environments with a primary goal of facilitating interactive learning at remote sites throughout the state.

The results of their study and experience demanded that their station equipment:

1. Be portable and self-contained;
2. Is easy to set up and operate;
3. Operate on 4-wire private line voice-grade telephone circuits;
4. Facilitates student participation in the classroom discussion i.e., interaction;
5. Have sufficient acoustic output power for classrooms seating up to 200 people;
6. Be a public address system for its own classroom;
7. Minimize acoustic and electrical noise;
8. Be reliable and easy to maintain.

Darome designed station equipment utilizing these guidelines. Darome, Inc. was able to respond to the design request because of circuits available from its line of modular sound equipment and prior experience in sound system design. The ETN station equipment, developed in two phases during 1973 and 1974, is commercially available today as a Darome Model 604 Convener.

To better understand the operation of the Model 604, imagine for a moment you are at the Walworth County Courthouse in Elkhorn, Wisconsin. This is the third in a series of four sessions on "Public Information for Clerical Staff of Social Agencies". Today you arrive early and observe Sally, the classroom facilitator, removing the Model 604 from the audio visual equipment room. She has no difficulty carrying the suitcase like unit to room 103 where the class begins in ten minutes.

She sets the unit on the table in front of the room, opens the unit's storage compartment, and removes the cables. As she plugs the AC cord into a nearby outlet, she notes that the indicator near the volume control is on indicating that there is power to the unit. Next, she plugs the unit's telephone cord into a receptacle on the wall. Background music originating from Madison is heard. The connection to Madison is part of a state-wide 4-wire leased line circuit with outlets in 172 similiar classrooms. Sally adjusts the single control for a comfortable music level within the classroom and proceeds to set up the microphones at the tables in the room. The twenty foot cords facilitate the positioning of one microphone on each of the four tables. The microphones are rugged cast metal with a heavy base and sit upright on the table for easy access by the 12-16 participants.

The music stops and the course is introduced. After the introduction the instructor greets several students at various points on the network. As they respond it offers reassurance of others in attendance and encourages interaction.

After a brief lecture the instructor pauses to answer questions. A person across the room reaches out for the microphone, presses down on the drawbar, and asks the first question of the evening. A portion of the signal, which is fed to the network, is amplified in the classroom enabling those across the room to hear the question as clearly as those at other network locations. This short lecture, question and answer format, continues for the remainder of the third session. After dismissal you leave the room satisfied with your remote learning experience. The system had been remarkably free of annoying sounds frequently encountered with public address systems and certain telephone calls.

You wonder why? For a better technical understanding of the Model 604, consider the functional Block Diagram shown in Figure 1. The receive channel has a balanced line input transformer. The input receive preamplifier is compensated to attenuate signals outside the telephone line bandpass. The signal is passed directly from the receive preamplifier through S2 to the receive volume control, and from there to the 14 watt RMS amplifier. The cabinet contains two 8 ohm speakers wired in parallel. After S2, the signal is also fed to a second preamplifier that drives signal detector circuits which controls the operation of S3.

The signal source for the transmit channel is a high quality low impedance cardioid microphone connected to a voice compensated transmit preamplifier. With S4 closed, the signal passes to a 20 db loss pad and switch S3.

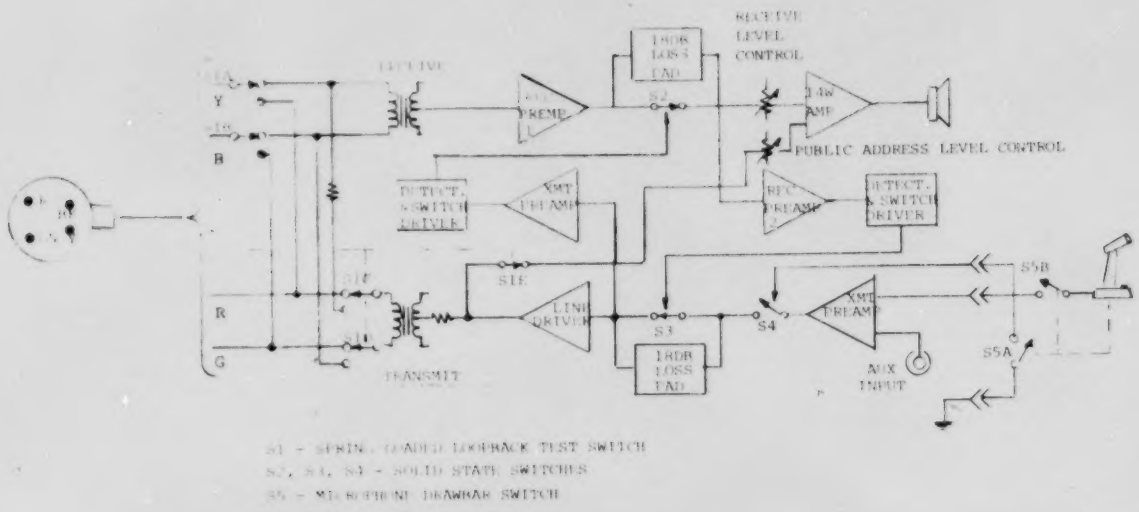


FIGURE 1 FUNCTIONAL BLOCK DIAGRAM MODELS 604, 611 CONVENER

With S3 closed the signal is fed unattenuated to the balanced output line driver. The signal from S3 also fed to a second transmit preamplifier that drives a detector circuit which controls the operation of S2.

To understand the design of the transmit channel, consider the potential acoustic problems with several hundred microphones tied to a common audio circuit. This dictated that the microphones be normally off. As seen from Figure 1, the internal wiring of the Model 431 microphone leaves the microphone element disconnected from the transmit preamplifier until the drawbar is depressed. This insures that the microphone in use does not activate the other microphones within the room. When the drawbar is depressed, S5B closes which in turn closes solid state switch S4. Having all microphones off eliminates audio pickup from the classrooms. Voice gated microphone operation was not used because the noise to be eliminated would operate the voice gate and pass the unwanted noise to the network.

Unwanted electrical noise on the network originates in the form of white noise at the highest gain point in the circuit. In this case it is the transmit microphone preamplifier. With potentially 175 conference units operating on an additive circuit the resulting white noise would prohibit any intelligible communication. For this reason, S4 is left in a normally open state blocking this white noise from all stations not participating in the discussion at a particular instant.

To achieve a good signal to noise ratio on the network the Model 604 must transmit a high quality signal. To achieve a good transmit signal, all the elements in a sound reinforcement system and their relative locations in the room, and the room acoustics must be considered. In Wisconsin's ETN classrooms, room acoustics had to be accepted for two reasons. The first being that in many cases the classroom facilities are not under control of the University, such as, those in hospitals, courthouses and municipal libraries. The second reason is financial. The cost of acoustic treatment for some 175 classrooms would be prohibitive.

The elements of the sound system include the microphone, amplifier, and speaker. These elements must be optimized to eliminate two acoustic problems commonly referred to as echo and barrel effect. To the person at a network location listening to another person speaking from a second location, the listener experiences the illusion that the talker is at the bottom of a barrel. This problem is aggrivated by live rooms. The live room has a high content of reflected sound because there are insufficient absorbtive surfaces to dissipate acoustic energy. Consequently, the acoustic energy decays slowly, or the room is said to have a long reverberation time. The incident, or direct sound, is transmitted followed by transmissions of the reflected sound. The reflected sound is delayed from the incident sound and thus creates the barrel effect. Basically, the barrel effect occurs in two situations. First, when the transmit preamplifier gain is increased so the speaker can be picked up across the room from the microphone. With the gain that high, the incident and reflected sounds will be transmitted creating the problem. The second occurs when the public address level is set too high. The talker's incident sound is picked up by the microphone and amplified to such a level that the reverberant sound from the Model 604 amplifier/speaker becomes a significant portion of the sound incident to the microphone.

Minimizing these problems begins by selecting a highly directional cardioid microphone. The cardioid microphone responds well to signals directly on the front access of the microphone and the response drops off sharply on the sides and rear.

A second step toward achieving a good signal to noise ratio is to fix the transmit preamplifier gain. The fixed gain requires that the talker be within 12 to 18 inches of the front of the microphone to transmit an acceptable signal level. This restriction eliminates the first cause of the barrel effect mentioned above. Another benefit is minimizing the transmission of unwanted room noises during the time a talker's microphone is on. To the remote network listener, the talker's voice is heard at a considerably higher level than the background noise within the room.

Consequently, good signal to noise ratio exists. Additionally, the frequency response of the transmit preamplifier is tailored for speech intelligibility and telephone line compatibility.

To control the second cause of the barrel effect the public address level is adjusted by a control separate from the receive level. The public address level can be set at an adequate level for the classroom. As the receive is adjusted for comfortable listening level, the public address level is maintained at its desireable setting.

Echo is a common problem in telephone circuits. Through via net loss design concepts it is contained to the point where the user is virtually unaware the problem exists. Echo is that portion of a transmitted signal which is retransmitted by the receiving end of the signal path. This signal is delayed in time by the transmission path. The station equipment on ETN contributes to echo when a signal is received, amplified in the classroom, and transmitted back through ETN by an open microphone. The delay encountered is that time required for the sound to travel from the speaker to the microphone within the classroom. This echo is eliminated with the Model 604 by inserting 40 db of loss in each loop on the network which is a potential source of echo. 20 db of this loss is in the talker's receive channel and 20 db in the listerner's transmit channel.

This is accomplished in the Darome Model 604 S2 and S3 as shown in Figure 1. The switching operates as follows: With the microphone switch depressed and a transmit signal present, a portion of the signal is fed to transmit preamplifier 2 which in turn attenuates the receive channel by 20 db, that is S2 opens. The figure of 20 db was arrived by considering the number of stations involved, a typical number of microphones that might be in operation at a given time, and the amount of delay expected. The amount of loss required is proporational to the circuit delays. With satellite communications this loss must increase to 60 db total. The 20 db loss in the receive channel inhibits network signals from being received, amplified in the room, and retransmitted by the open microphone. To further understand the operation, consider a unit at a station where the transmitted signal is received, that is a listener's location. The listener's receive channel has a signal sensor. With the signal present receive preamplifier 2, the detector opens switch S3 and inhibits the transmit channel by 20 db in the same way the receive channel is inhibited at the station where the signal is being transmitted from. In other words, if a student in a listening room were to depress his microphone and attempt to speak with a signal present on his receive channel, his outgoing signal would be attenuated by 20 db.

The final design requirement concerns the equipment reliability and maintenance. With the extensive statewide network encompassed by ETN reliable operation was mandatory. The design began by using the latest in integrated circuit, solid state technology. All of the electronic components subject to failure are contained on a single, epoxy, plug-in printed circuit board. A customer's service department can replace the defective board with a spare and return the defective board to the manufacturer for repair.

As an alternate step the conference unit itself can be tested by activating the same loop back test switch. This shorts the units transmit pair to its receive pair. By turning the receive level control to maximum volume and talking into a microphone, the signal is transmitted and received by the same unit. While the audio is distorted by the echo suppression gating, it does verify the units operational integrity. This loop back test switch is a spring return slide switch. This provides a simple operator presentation and maintains the unit in its normal mode of operation.

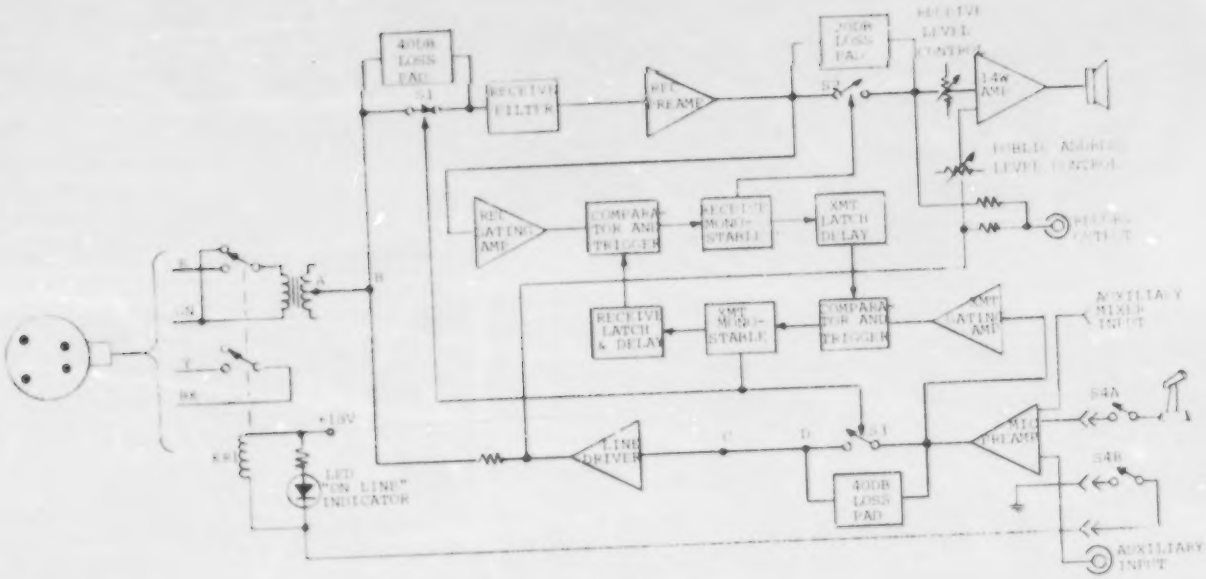
Two-Wire POTS Station Equipment

Two-wire station equipment used in conjunction with ETN to bring guest lecturers and instructors to the people of Wisconsin via the world wide dialed telephone network. Many of the design features applying to the 4-wire equipment apply to the 2-wire station equipment. Portability was a primary consideration since ICS ships the unit via common carrier to an off campus location for origination of an ETN program. Frequently the equipment is hand carried on commercial flights. The Model 680 carrying case which fits under a commercial airline coach seat, houses the Model 610 Convener, its basic 491 microphone and up to three additional Model 461 microphones.

To understand the operation of the two-wire station equipment consider the functional block diagram shown in Figure 2. The Model 610 is operated in conjunction with a standard telephone, which is used to dial outgoing calls and provide ringing for incoming calls. Transfer to the Convener is made by switch S5 on the Model 491 microphone. Operation of switch S5 closes relay KR1. This connects the tip and ring of the telephone line to the Model 610 and provides an auxilliary set of contact closures on the yellow and black leads of the telephone cord for A and A1 controls on key telephone equipment. As with the Model 604, the addition of a Model 603 mixer allows the use of four additional microphones.

With KR1 operated, an incoming signal passes through the balanced line transformer, normally closed S1, and the receive filter/receive preamp combination. If the signal is of sufficient amplitude, the preamp comparator and trigger circuit will pulse the monostable. The monostable output will close S2 and permit the signal to pass to the receive level control and then power amplifier. The monostable period is 100 milliseconds. When it is activated it latches the transmit comparator such that the transmit channel cannot be on simultaneously with the receive channel. The monostable is retriggerable. Any incoming signals of sufficient amplitude retrigger the monostable and reinitiate the 100 millisecond time period. Therefore, longest time the receive channel will remain on is 100 milliseconds after the last signal of sufficient amplitude to operate the gating. There is an additional delay in the order of milliseconds after the termination of the receive signal before a transmit signal can be initiated.

The transmit channel begins with a highly directional cardioid microphone similiar to the Model 604. The microphone preamp is voice compensated in the same manner as the Model 604's preamplifier. The signal from the microphone preamp feeds the transmit gating



S1,S2,S3 - SOLID STATE SWITCHES
S4 - MICROPHONE SWITCH
XX1 - PHONE TRANSFER DELAY

FIGURE 2 - FUNCTIONAL BLOCK DIAGRAM - MODEL 491, 610 CONSOLE

preamplifier. This preamplifier provides frequency response shaping and drive for the comparator and trigger circuit. If the signal from the gating preamplifier is of sufficient amplitude it will trigger the transmit monostable. The monostable drive closes switch S3 and opens switch S1. With S3 closed the transmit signal is passed to the line driver and then to the telephone line. The transmit monostable is retriggerable and operates like the receive channel with a maximum of 100 milliseconds hangover time. Again there is a delay of milliseconds before the transmit channel releases the receive comparator. Besides the telephone line, the line driver provides signal to the PA level control which feeds directly to the power amplifier bypassing any receive gating. Many of the acoustic and electrical noise mitigating techniques employed in the Model 604 are also part of the Model 610. The transmit channel gain is fixed and the high quality cardioid microphone minimizes the pickup of room acoustic noises during periods of transmission. Note that the microphone element itself is not connected to the preamp amplifier until the drawbar is depressed. Once the drawbar is depressed, the voice operated switching takes over. Again, the talker is forced to be within arms length reach of the microphone to depress the drawbar. The voice operated switching was designed to be effective within that range. As the talker's sound pressure level at the microphone continues to drop below the acceptable limit, the gating functions intermitently, and then not at all. Having switch S3 open eliminates the primary source of electrical white noise, the microphone preamplifier. Since each microphone contains its own S4 switching, only those microphones in use will be a potential source of ambient room noise.

The Model 610 in the quiescent mode attenuates telephone line noise by 20 db. This 20 db loss is switched out of the circuit when S2 closes. With the receive normally off, telephone line noise below the gating threshold is virtually not heard. Distractions caused by telephone line cross talk are minimized as it is normally heard only during the presence of a signal with sufficient amplitude to operate the voice switch. Therefore, the listener is not able to focus his attention on the noise.

Echo on the POTS station equipment is eliminated by the simplex operation. That is, Model 610 is either in the transmit or receive mode, and therefore, cannot transmit a station receive signal or receive a telephone circuit echo.

From a maintenance standpoint, the POTS station equipment is completely solid state with integrated circuits for all major circuit functions. For maintenance and test reasons, the circuitry is contained in a single plug-in printed circuit board. G10 glass epoxy printed circuit board material was chosen for strength against abuse portable equipment must endure.

An LED indicates when the Model 610 is actually connected to the telephone line. The unit left on is equivalent to an off hook condition on a normal telephone. The indicator serves as a reminder to turn off after call completion.

Again the simplified controls include the receive volume control and the screwdriver adjustment for the public address level. Only one microphone (Model 491) is equipped with a switch to answer the phone line. For simplicity, all the microphone jacks are the same, allowing the special microphone to be plugged into any of the four jacks.

The success of these design efforts are reflected in the technical portion of the ETN student evaluation forms. Now in its eleventh year ETN is serving the 175 stations. During the 1977-1978 school year 171 program sessions are being offered. This includes 1355 individual sessions totaling 1,796 hours of ETN programming. Taking the 186 days in a school year multiplying by 15 hours a day and adding 4 hours for each of the 36 Saturday mornings of the period totals

2,934 possible hours of programming. Recall that ETN is a gigantic party line circuit. Because of this, it is restricted to one simultaneous broadcast of course material at any given time. Further, consider that the current 1,796 hours, will be the prime hours of the possible 2,934 hours available. While the courses on ETN are offered to all 175 locations, because of special regional interests, not all of these locations will have students signed up for a given course. With greater network flexibility an additional course(s) could be offered simultaneously. One form this flexibility might take is to transfer the unused portions of ETN to secondary regional bridging. These regional private line bridges could be accessed through the dialed network and associated conferencing equipment. The University Extension has the teaching staff and the technical manpower at the ETN Control Center to provide simultaneous programming.

Consider for a moment the needs of conferencing in business. Conferencing is conducted between regional manufacturing facilities, between a main bank and branch banks, a home office marketing and/or sales department to its regional branch offices, etc. These conferences include such items as management coordination, production scheduling, new product training, loan meetings between the branch banks, etc. Because of the relatively limited nature of this conferencing, and the distances involved, generally it is not practical to consider a multipoint private line telephone network.

The evolving ETN system and business conferencing have a common need; to use the POTS Telephone Network to achieve flexible conferencing.

POTS Station Equipment Improvements

While the present 2-wire station equipment has been used successfully in both educational and business conferencing, experience has revealed areas for improvement. On point to point telephone calls level problems are more tolerable because of the ears adaptability to a wide range of signals. However during a conference call, persons exchanging information create a direct comparison of one persons level to another. The first area for improvement then would be to eliminate most of these obvious level differences.

On the Model 610 the first step towards achieving this would be to add compression in the transmit channel. The compression function would be inserted between points C and D on the Block Diagram in Figure 1. While the gating sensitivity is adequate for pick up of the average persons voice within arms reach of the microphone, there is a considerable level variation as the same person moves to within inches of the microphone. The preamplification would be added to achieve a more consistent level from those persons speaking at arms length of the microphone. The compression point would be set to establish 0 dbm transmit level peaks on the Model 610. Note the gating would remain at the microphone preamplifier output and additional gain and/or compression would not be added prior to the gating pick off point. If it were, the increased gain in the gating channel would lower the sound pressure level required to gate the transmit channel on thus transmitting more room noises. It is also not desireable to have the compressor function prior to the gating. This would cause a constant shifting of the gating sensitivity at the microphone. A loud talker for instance would compress the gain of the transmit channel and then a soft talker would not be able to gate the system on until the compressor released the required gain. Compression in the transmit would help equalize the level at the station equipment output. This leaves two major areas for possible level problems. In the telephone line connection itself, and in the bridging equipment.

To better understand typical line losses consider the simplified telephone connection illustrated in Figure 3. Signal loss from the telephone to the

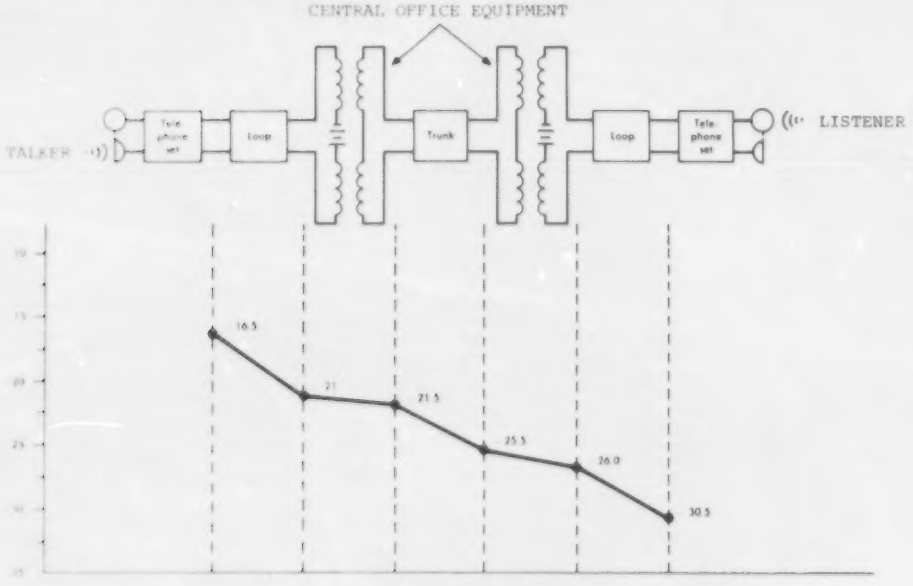


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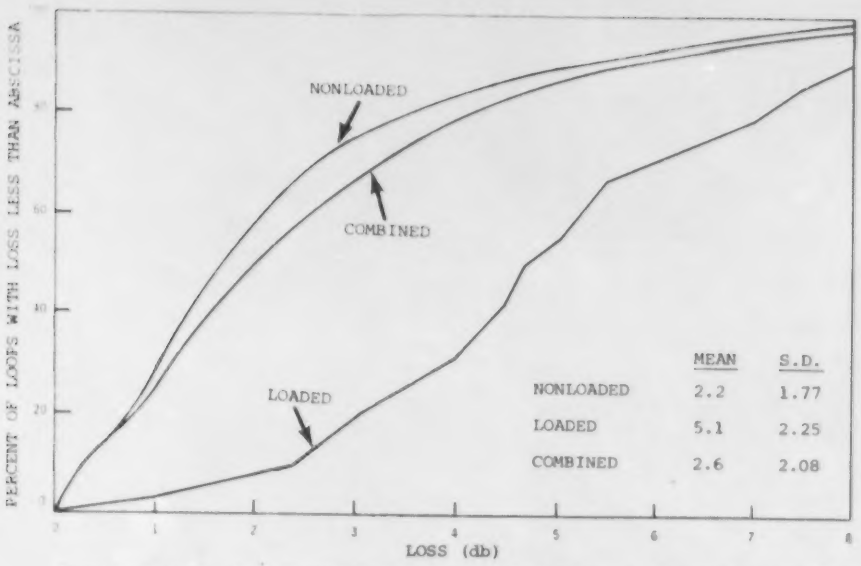
FIGURE 3 VOLTAGE LEVELS IN TYPICAL TELEPHONE CONNECTIONS



Reprinted from "DATA COMMUNICATIONS USING THE SWITCHED TELECOMMUNICATIONS" Bell System Technical Reference PUB 41005.

FIGURE 4 - 1000 Hz INSERTION OF LOOPS TO BUSINESS CUSTOMERS

central office is typically 4.5 db. The connection through the central office to the trunk circuit has a .5 db loss, 4 db loss in the trunk circuit, .5 db loss in the end office connection, and a 4.5 db loss from the end office to the called party's telephone.

Figure 4 illustrates the 1000 Hz insertion loss from the customers telephone to the end office. As shown, the combined loaded and unloaded signal lines have a mean loss of 2.6 db with a standard deviation of 2.8. For practical limits on level adjusting equipment consider two standard deviations from the loss limits. Two standard deviations would include 96% of all connections made. Using this criteria the customer equipment to end office connection would have a loss range from virtually 0 db to approximately 6.8 db.

Applying the same process to calls with a connection length of 180 to 725 miles (see Table I) and using the data between primary sites, the mean end office to end office loss would be 7.3 db with a standard deviation of 2.3 db. From mathematics, the standard deviation of the sum of two or more independent random functions is obtained by squaring the standard deviations of each of the functions then adding the squares and finally, taking the square root of the sum. Using this technique the 3 loss segments can be reduced into a single composite mean and standard deviation. Employing this technique results in a customer premise to customer premise mean loss of 13.5 db standard deviation of 4.58 db. Again using two standard deviations as the limit criteria this would yield a loss range from 4.34 db to 22.66 db. While the loss of 4.34 db is somewhat unrealistic for inter-office calls it could be number encountered in calls placed within the same exchange or terminating in the same central office.

All of the above is based on using only two standard deviations which in turn implies that one out of every 25 calls placed for conferencing would have to be replaced in order to get losses within the acceptable limits. Using three standard deviations the loss range would be from virtually no loss to a maximum of 27.24 db.

To achieve good conferencing these level variations must be minimized, if not eliminated. Where special station equipment is used for conferencing it would be practical to consider a unit with a self-test feature operated in conjunction with a micro-

TABLE I - COMPARISON OF CONNECTION LOSSES AT 1000-Hz FROM 1966 AND 1969/70 SURVEYS

Connection Length (airline miles)	1969/70 Survey				1966 Survey	
	Primary		Secondary		DDD	
	Mean (dB)	S.D. (dB)	Mean (dB)	S.D. (dB)	Mean (dB)	S.D. (dB)
All	6.7 ± 0.6	2.1	6.6 ± 0.3	2.1	---	---
0-180	6.5 ± 0.7	2.0	6.4 ± 0.4	2.1	7.0 ± 0.4	2.3
180-725	7.3 ± 0.4	2.3	7.1 ± 0.6	2.1	8.5 ± 0.6	2.5
725-2400	7.7 ± 0.5	2.5	7.3 ± 0.3	2.0	8.9 ± 0.6	3.0

processor controlled bridge to compensate these levels by actual tests, measurements, and adjustments.

One possible mechanization for this idea is shown in Figure 5. The functions between points A and B in Figure 5 would insert into the block diagram of Figure 2 at corresponding points A and B. After initiating an outgoing call or answering an incoming call the timing sequencer would be set to start. The test signal originated by the bridge would start the test cycle. The test signal bandpass filter would insert 14 db of gain prior to measuring the incoming test signal to compensate for the typical incoming telephone line loss. The bandpass filter output would pass to a peak detector and on to the sample and hold where the peak incoming signal level would be held as a reference for the level programmable oscillator. At the termination of the input test signal the timing sequencer would send an outgoing test signal from the level programmable oscillator. The signal, fed to the telephone line by placing S1 in position 2, would reflect directly the incoming test signal from the bridge. Next the timing sequencer would shift the level programming reference for the sample and hold circuit to a 0 dbm transmit level. Level programmable oscillator would then output 0 dbm to the line at the station equipment. The second signal is measured at the bridge. The first measurement gives the entire loop loss of the connection. The second measurement gives the loss from the station equipment to the bridge. By taking the difference in the two signal levels the loss in both directions of the connection would be available at the bridge. This information could then be used to set the appropriate gains in the receive and transmit legs of the bridge to equalize levels at the bridge. Telephone lines with loop loss in excess of predetermined limits would be rejected and have to be replaced. A simple noise test could also be incorporated at both ends of the connection. Lines would be rejected for excessive noise.

With the increasing availability of complex functions in integrated circuits, these techniques are not only possible but will be practical for people doing extensive telephone conferencing.

One of the more cumbersome aspects of this type of teleconferencing equipment is the microphone and its associate cabling required to facilitate participation. While the setup is relatively simple, the cords present a rather awkward site and inhibit the flow of traffic within a conference room.

Within the past year, Senheizer of Germany has demonstrated, and made available to the consumer market, stereo headphones for use with a home hi-fi system using wireless techniques. The main hi-fi receiver unit has an infrared transmitter with the infrared detectors attached to the headset. This enables the hi-fi enthusiast to be anywhere within the room and listen privately. The technique is relatively insensitive to direction and signal blockage even by placing a hand over the detector area. A reverse application of this technique to the station equipment would certainly facilitate setup and provide greater flexibility for student participation since the microphone could be passed around the room.

A second technique considered by Darome is to have a single microphone with a long cord reaching from the conference or classroom table to the convener itself. From this microphone additional microphones could be looped to the first one using three to four foot cords. This would make a single cable run from the table to the convener with a minimum amount of cable on the table itself.

A third possibility would be to combine the first infrared transmitting technique with the second. That is, have a single microphone on the table with the infrared transmitter and the other microphones on the table plug into that unit. The infrared transmitting microphone could have a rechargeable battery and the convener have the charging equipment.

These speculative improvements are not necessarily part of current engineering projects at Darome. They are presented to stimulate ideas on technical improvements for telephone conferencing.

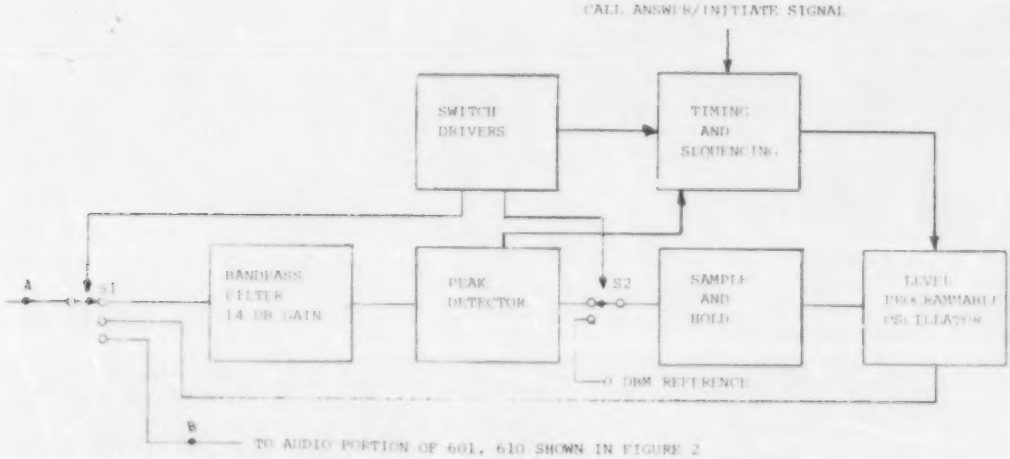


FIGURE 5 FUNCTIONAL BLOCK DIAGRAM LINE TESTING AND LEVEL ALIGNMENT

Bibliography

American Telephone and Telegraph, Technical Reference: Data Communications Using the Switched Telecommunications Network, May 1971.

American Telephone and Telegraph, Technical Reference: 1969-70, Switched Telecommunications Network Connection Survey, April, 1971.

Bell Telephone Laboratories, Incorporated, Transmission Systems for Communications, Fourth Edition.

Braun, D.P., Gilbertson, D., Hansen, H.C., "ETN: A Technical System". The Status of the Telephone in Education Second Annual International Communications Conference, University of Wisconsin, Madison 1976, pp. 51-74.

Clement, M.A., "Transmission", reprinted from Telephony, Telephony Publishing Corporation, Chicago, Illinois, February 1973.

Parker, Lorne A., "Teleconferencing as an Educational Medium: A Ten Year Perspective From the University of Wisconsin - Extension". The Status of the Telephone in Education Second Annual International Communications Conference, University of Wisconsin, Madison 1976, pp. 1-26.

Seaman, Millie, Interoffice memo to Lorne A. Parker, University of Wisconsin - Extension, Unpublished, March 29, 1977.

ICS PRODUCTION SERVICES:

AN OVERVIEW

By

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Early programming over Wisconsin's ETN, the Educational Telephone Network, consisted primarily of lectures delivered by University professors, followed by question and answer sessions. The need for a great deal of support in the audio "production" area was minimal at best. While a number of presentations were pre-recorded, with the instructor available for questions at the conclusion of the pre-recorded material, the facilities required for this service were fairly basic--a small recording studio with one or two microphones, a small mixing console, and a tape recorder.

However, 10 years of ETN programming have provided instructors additional insights into effective methods of presenting material over an audio network. Lecture material is supplemented with pre-recorded examples illustrating important concepts. Social workers not only are told how to deal with problem situations, but also listen to various short dramatic "actuality inserts" of what might occur in a given situation. Music teachers not only hear of new approaches in teaching, but also listen to recorded or live examples of musical techniques.

The growing sophistication of ETN programming necessitated an updated and more complete production facility. The simple one or two microphone studio was no longer adequate to meet the needs of programmers. Production facilities needed to include the capability of mixing several microphone tape and record sources into one, two, or more channels simultaneously. Studios needed to accommodate several microphones, and several studios were necessary to provide simultaneous programming and production activities.

The present ETN studio facility is located in Old Radio Hall on the Madison campus. It consists of a master control room (Studio A control) and five studios, four of which are in Radio Hall. The fifth (Studio M) is located in the Medical Library on campus. Studio A, the largest and the main production area adjoins the A control and network control rooms. Studio B and C are on either side of the network control room, but not visible from A control. Studio S, the SEEN (Statewide Extension Education Network) studio, is not visible to either Studio A or network control.

Up until the summer of 1977, many of the production facilities in use were inherited from Extension's WHA Radio, which previously occupied the building. This included all studio wiring (microphones, speakers, etc.), control room wiring and physical layout, and mixing console (monophonic, nine-channel built by WHA personnel circa 1960).

Due to age, many components began to fail regularly. Wire installation was decaying, causing short circuiting, and console preamplifies were tube type, occupying considerable space with frayed connecting cable. Rather than repair an already obsolete system, it was decided to start from scratch.

The first phase began during the summer of 1976 with the rewiring of Studio A, the main production studio. Eight 8" x 10" holes were drilled in the studio floor to accommodate floor boxes which house production mike jacks, ETN Edu-Com mike jacks, production headphone jacks, and other network operations related inputs and outputs. The floor boxes are recessed approximately six inches, with their hinged covers flush with the studio floor. The installation of these boxes eliminated several long cable runs over the floor space which were trip hazards in the previous arrangement. Since the holes were drilled through the floor, any noise from the level below the studio was easily heard. To dampen this, each hole has an outer metal box into which the box containing the various jacks is inserted. This acts as an additional sound lock. The floor box containing the various jacks is not permanently fastened to the floor, but rather rests on one inch flanges. This facilitates easy removal of any box for maintenance or additional wiring. Studio A is now wired to accept up to eight balanced production mikes, with expansion provided for up to sixteen. Eight monophonic headphone jacks are fed via the mixing console using half of a two-watt stereo amplifier with a nominal output impedance of eight ohms. Monitoring is provided by two Electro-Voice Sentry II-A monitor speakers fed via the console from a 20 watt RMS stereo amplifier. The control-room-to-studio talkback function is accomplished via a push-to-talk mike on the console fed to the other half of the two watt stereo amplifier which supplies a small, wall-mounted speaker in the studio.

Specifications for a new production console were developed, the console ordered, and installation completed during the summer of 1977. The console is of a 16 input x 4 output configuration, with monitor facilities capable of selecting any or all outputs, foldback, any input, solo, mono, or stereo tape. The entire system is wired for balanced 600 ohm operation.

ETN is a monophonic (single channel) system, which raises the question of need for a four-channel console. Flexibility is the primary reason. The four output busses are not utilized to feed a four track tape deck. Rather, they may be used as distribution channels for various programming activities, for example: to supply music to the SCA (FM subcarrier) system, feed ETN or

SEEN operations with various program material, or even act as a back-up to the master network control room.

All console inputs (high level only) and outputs are accessible via the A Control patch panel, allowing for any usage configuration desired. Also patchable are studio monitor inputs and outputs, tape deck inputs and outputs, turntable pre-amps, talkback, foldback, equalizers and filters.

One of the more convenient features is a shielded 11 pair cable from A Control to the master network control room. Six of the 11 pair are used as studio loops appearing on patch panels in both rooms. Additional cables also are run from network control to the SEEN auxiliary control room, and eventually to the faculty production area. By means of using the appropriate patch routes, Studio A control can feed or receive material from any control point in the building. The remaining five pairs of the 11 pair cables are used as remote studio and Direct Distance Dial (DDD) transmit and receive lines. These are also available at the patch panel.

A Control utilizes three amplex 440 series tape decks, each strapped for 3 3/4 or 7 1/2 ips operation. Two machines feature record and reproduce capability, with the third being a reproducer only. Track formats include full track mono, record and reproduce, two track stereo record and reproduce, a half track mono record and reproduce, and one-fourth track mono or stereo reproduce. Also available is a Gates Criteria broadcast cart machine equipped with record and reproduce electronics. Each reel to reel tape deck may be operated locally, or via remote control available at the production console.

Since Studio A Control is capable of acting as a control center for Studios B, C, SEEN, and remote Studio M, it was necessary to provide a talkback function to all of these points. This is accomplished by bringing the talkback line level output up to the patch panel, where it is normalled to the monitor amplifier input which serves Studio A. With the talkback output available at the patch panel, it may be routed to any studio desired, or via the use of multiples, to several studios.

Additional components include an audio filter unit and an additional audio equalizer, inputs and outputs being available at the patch panel. Primary audio equalization and filtering is accomplished at the console. Eight input modules offer an extensive range of frequencies in the low (40-160 hz) middle low (200-900 hz) middle high (1.5k-7.2k) and high (10-15 k) ranges with plus or minus 12db possible. Any of the 16 inputs is wired to accept identical input modules with associated EQ. This configuration allows individual tailoring of each input, rather than being confined to equalizing a master output, thus affecting all inputs fed to that channel. Each input is also wired to accept either a high or low level source, or both, making 32 inputs available, although only 16 may be used at once.

Studios

While Studio A serves as the primary production room, four others are available. Studios B and C, which are mainly used for ETN programming, are much smaller (approximately 10' by 15') and comfortably seat up to four individuals. Mike mixing for these studios is initially done in the network control room via Shure M-67 mixers. Each studio is wired to accept up to four microphones, and jacks are wall mounted. The output from the M-67 is fed to the network control patch panel where it is available for routing to Studio A via the previously mentioned studio loops.

Studio S is the main SEEN studio classroom, seating up to fifteen. Four mike jacks are wall mounted and fed to a small eight input, two output mixer in the SEEN auxiliary control room. The desired output from the patch panel in this room may be routed to network control, and then to Studio A via patching.

A remote studio located in the Medical Library on campus is also available. This studio is equipped with two AKG224E condenser microphones fed to an M-67 mixer. The line level output from the mixer is fed to telephone lines which appear in network control located in Radio Hall. A wall monitor speaker in Studio M is fed by a small one-half watt amplifier. The signal to the amplifier is supplied from Radio Hall via another telephone pair. Both transmit and receive paths of Studio M appear on the Studio A control patch panel via the interconnecting studio loops to network control. The phone lines used are unconditioned voice grade circuits. They provide a suitable bandpass for voice applications, and at the same time are available at relatively low cost. Studio M is used only for voice recordings or programs.

Recording Procedures

All recordings for use on the ETN or SEEN networks are recorded at 7 1/2 ips, half-track mono. This conforms with the available formats in the network control room.

The choice of 7 1/2 ips versus 3 3/4 ips is based not on the parameter of frequence response, but on ease of handling. Since the bandpass of a conventional telephone circuit is restricted from 300-3 khz, and since much production work involves voice only, 3 3/4 ips would certainly provide adequate fidelity as might 1 7/8 ips. However, we have standardized on 7 1/2 for two reasons. Many of our clientele have had very little exposure to sitting in front of a microphone. For many, it is a first-time experience. As a result, they often feel ill at ease, and mistakes can be rather numerous, requiring considerable editing. Since 7 1/2 ips is a much more manageable speed at which to edit, its use speeds up post production work considerably.

Secondly, many prerecorded segments are distributed in cassette form via independent study courses. In these cases, the fidelity provided by a higher tape speed becomes more important.

Selection of a recording tape should be based on the intended use of the tape. For all network program production voice recordings, Scotch 176 and 177 are used. Scotch 176 is a 1 1/2 mil. tape while 177 is 1 mil. Both are good voice tapes. No tape thinner than 1 mil. is used due to its susceptibility to stretching (if tensilized polyester) and increased possibility of print through. Both 176 and 177 have identical optimum bias settings and may be interchanged with no difficulty. Where a more critical tape is called for, such as in music mastering, Ampex 406 or 407 is used (production only). They require a slightly higher bias setting which is easily accomplished on the Ampex 440 series tape deck. Ampex 407 is also used for duplication mastering. While it provides a 2 1/2 to 3 db higher output and more overdrive capability, the most important parameter for high speed duplication is its oxide binder, which stands up well to the many passes required during a duplicating run.

Recording tape is available in a number of packaging modes. The most inexpensive method is referred to as bulk. In this form, tape is shipped either in pancakes wound onto a hub with no flanges, or wound onto a seven or ten inch reel, but with no box. In pancake form (available in 10 1/2 or larger reels only) if a 7 1/2 inch reel is required, it is necessary to buy empty reels and wind the bulk tape onto them. This requires additional operator time. We have found that the most economical route is to buy bulk tape wound onto 7 1/2 inch reels, but with no box. Boxes may be purchased separately. This saves anywhere from $1.50 - $2.00 per reel of tape if purchased in the manufacturer's own box and shrink wrapped.

Cassettes likewise are available in a number of packaging formats. Again, the most economical form is bulk. Cassettes are usually available in standard lengths of C-30, C-45, C-60, C-90, or C-120, but other lengths are also available. Bulk cassettes are identical to packaged cassettes (same tape, cassette shell, etc.) except they are shipped in trays of 25 or 100 with no sleeve or lable. Labels and various cassette sleeves may be purchased separately. Buying bulk realizes a savings of 20-50 cents per cassette. Since we apply our own labels anyway, the bulk cassette bought without a label is actually perferred.

The volume of cassette production at ICS has grown from a few hundred in 1974, to 15-20 thousand in 1977. In addition, over 1,000 reel masters of ETN programs are recorded annually.

Peripheral Services

Instructional Communications Systems offer cassette tapes of any ETN or SEEN program upon request. All network programs are recorded on reel-to-reel tapes from which cassette or reel-to-reel duplicates are made. In the event a student was unable to attend a session because of a technical or programming difficulty, tape copies of that session are provided at no charge, to be returned within 30 days. Requests for program tapes are processed and shipped in no more than two days, and in most cases, within 24 hours.

Duplicating facilities for use in network program dubbing are of the "in-cassette" type. That is to say, a cassette of predetermined length is inserted in the duplicator and a copy is made. The system runs at 12 times normal speed, with master transport speeds of 45 or 90 ips, corresponding to 3 3/4 or 7 1/2 ips and a slave speed of 22 1/2 ips. This translates to a C-60 in 2 1/2 minutes. Components consist of a master electronics assembly, master transport, a four bay slave assembly, and a three bay slave assembly with a master cassette bay. It is possible to duplicate from reel to cassette, cassette to cassette, or cassette to reel. Up to seven cassettes may be duplicated simultaneously from either a reel or cassette master. The track format is 1/2 track monaural. Both channels may be copied together or individually.

Other technical features of the system include: duplicator VU meter readouts of audio or bias signals, switch selectable, accommodation of reel sizes up to 10 1/2 inches, and with properly prepared masters, automatic shut off at end of master program or at beginning of program after rewinding.

Selection of the cassette shell and tape become more important at duplicating speeds. We have had the most success with shells that feature guide rollers mounted on lubricated pins to help reduce wow and flutter readings. We have also found that pressure pads mounted on metalic springs rather than on the tape shield seem to provide for better tape to head contact at high speed.

Tape parameters of importance include: 1) binding characteristics of the oxide to the backing - it must withstand high speeds without shedding or flaking. A "dirty" tape clogs head and accummulates on pinches rollers. 2) Tensile strength - the tape should not stretch or break while running or when stopping at the end of the cassette. Also important is a sturdy connection of the leader to the hub.

When duplicating more than seven cassettes, the system operation would proceed as follows. The master reel-to-reel tape is prepared with one inch foil strips at the beginning and end of the program. Cassettes are then loaded into the duplicator. The master tape is cued to the first foil strip placed at the prescribed distance on the tape from the program start. The master transport is started and after a delay of three seconds (to allow the master transport to attain operating speed), the cassette bays begin running and the duplication process begins. By placing the foil strip the correct distance from the start of the master program, it becomes possible to synchronize program start with cassette start. This means that the program will begin within one to three seconds after the leader on the duplicated cassette.

After the second foil strip is sensed at the end of the master program, the machine stops. The duplicated cassettes are removed, and more blanks are inserted. The operator then rewinds the master tape. After the foil strip at program start is sensed in the rewind mode, the machine again stops. The operator then may recue and begin another duplication run. The advantage of foil sensing is that it eliminates the need for constant rethreading, which slows down production.

Cassette Packaging

While many ETN programmers began requesting increased support in the audio production area, others began to take the teleconference idea one step further. The concept centered around duplicating a program series onto audio cassettes, and combining it with printed materials, usually in notebook form. These then couid be distributed to clientele not only in Wisconsin but anywhere in the countiry. They might be used as independent study programs or as reference material. This concept means that a program's potential audience need not be limited to ETN locations in Wisconsin, but may be expanded nationwide via tape recording, in this case audio cassette. In order to meet the demand for this type of service, ICS investigated the possibility of providing a cassette duplicating packaging facility to ETN programmers and other Extension personnel.

After carefully evaluating what would be required to begin a facility of this type, we concluded that several criteria had to be met:

1) The service must be a complete, one stop facility capable of handling a project from start to finish. The facility must provide for notebook supply in various formats. The notebooks must be customized to the clients' needs. This would include proper number of cassette cavities, spine and ring size, and silkscreening. Notebook mailing cartons must also be available.

77

A printing facility for cassette labels needed to provide several different types of styles, as well as point size. Color also needed to be available in any shade requested.

Finally, the facility needed to act as a central packaging shipping point for incoming orders, if requested by the client.

2) Cost must not be prohibitive, preventing some departments from using the service.

3) Technical characteristics of the duplicating facility needed to provide exact length cassettes (with program material the same length as the cassette to eliminate rewinding or fast forwarding to find the beginning of a side). The facility must also meet rigid performance specifications.

ICS began the service on a limited basis in 1976. Contracts were established with vendors to provide the hardware which could not, because of cost, be manufactured in house. These include customized notebooks and mailing cartons, as well as label printing. Through the University State Printing office, labels may be customized in any manner desired, including any colors, or color combinations, or art work requested.

Cassette duplication was limited up to 200 copies per master and was still of the "in cassette" method using the existing duplicating equipment.

The service was well accepted, prompting establishment of a full scale facility. However, the major weak spot involved cassette mass duplication.

"In cassette" duplication, as described earlier, is one method, but it involves a number of drawbacks. The cassette cannot be tailored to meet the exact length of the program. Usually anywhere from 30 seconds to a minute of the tape remains at the end of side one, or the beginning of side two of a duplicated cassette. This is necessary to allow for variations in the actual amount of tape wound into a cassette by the manufacturer. A difference of up to five feet is common. At cassette speed, this translates to approximately 30 seconds.

Additionally, duplication in cassette suffers technically. Wow and flutter parameters often become objectionable simply due to the physical limitations of the cassette. Also, each duplication bay is essentially an individual recorder. This results in slightly different characteristics from bay to bay causing occasionally variations in recorded sound noticeable to the ear.

The problem of "in cassette" duplication are circumvented via reel-to-reel duplication onto "pancake" cassette tape. This is the format which is now used. The facility consists of master electronics, a master transport, and three reel-to-reel slaves. Duplication is at 12 times normal speed, and is fully automated. System operation is as follows: The master tape again is prepared with the appropriate foil strips at program beginning and end. Instead of just one cut of the program on the master reel, up to six are recorded. This eliminates the need to rewind after just one cut. Bulk pancakes of cassette tape are loaded onto the reel-to-reel slaves. To begin duplication, the operator starts the master transport, which simultaneously starts the slaves. Duplication continues until the second foil is sensed. The master transport slaves stop, and the master automatically rewinds, stopping at the first foil strip. The system then restarts itself for the next duplicating run. The operator need only monitor the operation and reload the slaves when empty.

After the pancake reels have been duplicated, they must be loaded into C-0 cassettes. (A C-0 is a leader cassette.) This is accomplished by means of an automatic cassette loader which winds the pancake tape into the cassette, and performs the necessary leader-to-tape slices at head and tail automatically. A cue tone from the master tape is detected which initiates the loaders tape cutting and splicing sequence. After the cassette has been loaded, it is automatically ejected, and another cassette is put into position for loading. The entire loading operation is automatic. The operator need only monitor the system and re-supply the cassette feed magazine which will accommodate up to 45 C-0's. In addition, the loader will load cassettes with blank tape to any length desired. The entire duplicating and loading operation will accommodate tape thicknesses up to and including that used in a C-120.

The advantages to this method of duplication are numerous. Cassette length is exactly tailored to program length, thus wasting no tape. The technical characteristics of reel-to-reel duplication are superior to "in cassette" duplication, with wow and flutter less than .15% as compared with .2% or greater in cassette.

Perhaps the most important parameter is that of cost per cassette. Whereas a bulk C-60 may sell for anywhere from 50-80 cents, depending on type and vendor, a C-60, locally manufactured via the method outlined, amounts to 25-30 cents, depending on type of tape and cassette shell used. This cost savings is then passed on to the customer.

In addition, we have control over cassette length when loading blank tape. Instructional Communications Systems now offers a complete recording and packaging service. Clientele may do an entire project from initial recording to final packaging and shipping, or any phase of a project.

Interestingly enough, due to the popularity of the cassette packaging service now available to ETN programmers, other departments have shown an interest in pursuing this form of instruction. A number of independent study courses not related to ETN activities have been completed. This it seems that the growth of ETN has helped to produce a previously untapped support facility for the entire campus.

ADDITIONAL SERVICES

An informational system which is gaining popularity is referred to as "Dial Access." In most cases, it is a one-way system, with information passed on to the caller, but some systems do offer the opportunity to interact after hearing a pre-recorded message. Operation is rather simple. Messages are recorded onto cassette or cartridge, and when caller dials the system number and requests a specific tape, that tape is played by a tape deck interfaced with the phone line.

The choice of tape format cassette, cartridge or reel-to-reel depends largely on the expected volume of the system, estimated production and installation costs and so on.

Instructional Communications Systems distributes pre-recorded messages to the statewide University Dial Access System in cassette form. Cassettes are used rather than cartridges primarily due to their small physical size, which allows more messages per storage space. Also, the playback decks themselves are physically smaller.

Although Dial Access by its very nature uses voice grade telephone circuits and is limited in its bandpass, the recordings are professionally prepared to provide the best possible reproduction.

Dial Access master recordings are done in Studio A using the complete facilities available. They are done at 7 1/2 ips, full or half track mono. From the finished master, another reel-to-reel sub-master is prepared from which cassettes are dubbed. The sub-master is recorded dual track, with the message on track one in the forward direction, and track two in the backward direction. This enables both tracks to be duplicated at once, cutting production time in half.

Dial Access cassettes are recorded half track mono, cut to exact program length, with the same message on both sides. This eliminates the need for rewind or fast forward at the end of a message. All the operator needs to do is flip the cassette over, and it is ready for the next call. When tailoring a recorded cassette, the leader is also shortened by about half. This means that the message will start approximately two to four seconds after the cassette is put into the playback deck. This helps keep the time a caller must wait till the message begins down to a minimum. (This becomes more important when an incoming call is a toll call.) Cutting the leader too short may decrease the wait time, but it also causes additional difficulties. Frequently, the splice will be located near or on the capstan. If the playback deck does not have an automatic shut-off feature, the friction of the capstan and pressure roller may break the splice.

FACULTY PRODUCTION

Instructional Communications Systems offer an additional production facility for use by Extension faculty and staff. The faculty production room is a user operated facilty equipped with all components necessary for producing an audio presentation. The concept is based on the premise the individuals may have some tape projects which, for various reasons, do not require the services of an engineer or a professional audio studio.

Equipment includes two reel-to-reel decks with reproduce and record formats of 1/2 track mono, 1/4 track mono, or 1/4 track stereo, and tape speeds of 7 1/2, 3 3/4 or 1 7/8 ips. Two cassette decks with formats of 1/2 track mono and 1/4 track mono or stereo, turntable, and capability of recording telephone conversations are also included.

A small mixing facility provides eight input channels and high and low frequency shelf equalization. All inputs and outputs appear on a patch panel in the room. Complete instructions are provided, and an ICS engineer is available if difficulties occur. The facility is free of charge, with the exception of audio tape if required.

An additional production facility, referred to as the "You-Tape-It" studio, is also available. This system includes a professional reel-to-reel tape deck with 1/2 track mono format, and speeds of 3 3/4 or 7 1/2 ips. The tape deck is located in an adjoining room, visible from the studio via a glass sound lock. The "You-Tape-It" studio accommodates up to two people. A single AKG224E microphone is used, center mounted on the table via a gooseneck. The studio is entirely user operated and is available at no cost, except for recording tape if required. The tape deck is operated via remote control from the studio. A small column speaker in the studio provides playback monitoring, enabling the user to stop and listen at any point. The speaker is muted in the record mode.

In summary, the ICS production facility has grown in direct proportion to the programming demands of ETN. The cassette mass duplication and packaging facility were only farfetched ideas two years ago. Today, they provide a needed service to ETN programmers. As Dial Access facilities grow and are refine, the production department will need to meet the demands here also. In short, the directions pursued in the future will be dictated by the growth of the teleconferencing idea.

VISUAL SYSTEMS FOR TELECONFERENCING: TELEWRITING, TELEVIDEO, AND FACSIMILE

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Introduction

Teleconferencing is people communicating with people. In education, business, industry, health, and a host of other fields, telephone networks link individuals in many geographic locations. Communication, however, sometimes requires both audio and visual elements. Although teleconferencing has proven to be an effective medium for verbal interaction, it has been limited in its capacity to provide visual information.

New telewriting, televideo, and facsimile systems have recently been introduced that are reshaping our conception of the telephone. Often equated with only audio communication, the telephone is now becoming an important visual tool capable of transmitting a variety of graphic and pictorial information, from free-hand drawings and computer graphics to pictures and paper copies. This increased visual capability greatly extends the role of teleconferencing in meeting informational needs. In education, for example, many types of visual illustrations can supplement audio instruction. Business teleconferences can include charts and pictures to facilitate decisions. Industries may exchange written information, make graphic presentations, or construct computer drawn diagrams. Remote diagnoses are possible in medicine, where doctors simultaneously view x-rays and discuss patient symptoms.

Telewriting, televideo, and facsimile are not new concepts. The first telewriter was marketed nearly 15 years ago. A prototype televideo system was developed in the late 1950's, and facsimile became feasible in the mid-1960's when AT&T opened its lines for data transmission. What is new about these systems is their increased performance and versatility. In these respects, they bear little resemblance to their predecessors. Most of this progress is directly attributable to electronic-digital technology. Its application has not only improved existing products like electro-mechanical pens, slow scan televideo, and fascimile. It has also made other devices possible--the Electronic Blackboard, graphics tablets, light pen-video writers, and computer graphics systems.

The development of these visual systems is primarily the result of a concentrated effort to more fully exploit the informational capacity of the telephone and other narrowband channels. This effort has been fueled by three factors:

- The increasing need for organizations and individuals to exchange many kinds of information quickly over long distances.
- Narrowband media, such as the telephone and SCA-FM radio, make more efficient use of available communications channels.
- The telephone is widely available, interconnecting almost every home and business in the nation.

Telecommunications is a mushrooming industry, especially since the advent of satellites; and communications channels must be carefully allotted in order to accommodate all of the users who wish to transmit audio and visual information, alphanumeric copy, and computer data (Gundlach, 1976; "Telecom Changes," 1977). While broadcast channels transmit many kinds of information, including animated audio, they require wide bandwidth and cannot be made available to everyone. Coaxial cable and microwave channels also have wideband capacity, but their installation is expensive and they serve limited geographical areas. The telephone and other narrowband channels are therefore unique in their ability to efficiently and economically transmit information to people in almost any location.

Approach of This Paper

This paper reviews telewriting, televideo, and facsimile systems that can be used to transmit visual information over a narrowband telecommunications channel, particularly voice-grade telephone lines. The general operation of each system is described, and a number of products in that category are summarized. The products listed are only examples of those available on the commercial market. The electronics industry is very large, and new products are continuously being introduced. In addition, many firms develop systems specifically to meet a customer's needs; and other companies sell only the memory chips, graphic boards, circuitry, and other components that are the basic building blocks of a videographics system.

The systems are compared according to key evaluational considerations, but no attempt is made to recommend one type of system or product over another. As will be seen, there are often necessary compromises especially between cost and performance, and a "best system" ultimately depends on the needs of the user.

All of the equipment reviewed employs solid-state circuitry in varying degrees to enhance performance and exploit the limitations of a narrowband channel. Because of the importance of electronic technology in developing improved visual systems, some recent advancements in this field are discussed.

Data compression techniques are the subject of much research, and they may make it possible to transmit higher quality images in less time than is currently possible. Some of these techniques are summarized at the end of the paper.

Before describing each system in more detail, it may be beneficial to provide some background information. This includes an overview of the types of information these systems can transmit, the limitations

of a narrowband channel, and some important considerations in evaluating any of the visual systems.

Overview of the Visual Systems

Telewriters, televideo and facsimile are all methods of transmitting visual information. They differ, however, in the kinds of material that can be produced. These differences are summarized in Table 1.

Telewriters are capable of generating hand-drawn graphics--line drawings, simple diagrams, words, equations. The graphic information is produced by writing on an electronic surface with a special pen. As the writing and drawing is presented, it is instantly transmitted to remote locations and usually shown on a TV monitor. Telewriting systems include electro-mechanical pens, electronic blackboards, light pen-video writers, and graphic tablets. These devices can be applied to a variety of informational needs. Electro-mechanical pens are widely used in business,

TABLE 1

VISUAL CAPABILITIES OF TELEWRITING, TELEVIDEO AND FASCIMILE SYSTEMS

Capability	Examples	Device
Hand-drawn graphics	Writing Drawings Outlines Equations Diagrams Graphs	Telewriters Electro-mechanical pens Electronic blackboard Light pen-video writers Graphics tablets
Instant transmission Shown on monitors or permanent paper and acetate copies Stored on audio tape		
Hand-drawn graphics and computer-drawn graphics	Free-hand drawings Traced diagrams Computer drawn symbols diagrams, graphics, schematics, charts Alphanumerics Graphics functions (zoom, reduce, lable)	Computer graphics
Instant or delayed transmission Shown on monitors Stored on tape, discs or in memory Reproduced on printers and plotters		
Graphics and pictures	Hand-drawn graphics Prepared graphics Writing Monochrome pictures Color pictures	Slow scan televideo
Delayed transmission Shown on monitors Stored on audio tape Reproduced on paper		
Paper reproductions	Typewritten pages Documents Prepared graphics Pictures	Facsimile
Delayed transmission Permanent paper copies		

industry, sales, banking, insurance, transportation, and government for both multi-point and point-to-point communications. In education, they have been employed to teach home-bound youth and distant adult learners. The Electronic Blackboard, graphic tablets, and light pens are relatively new, but they are primarily geared toward education or business and industry. In most cases, both audio and graphic information can be recorded on regular audio tape for playback at a later time.

Computer graphics systems can be used for hand-drawn information, but they are more specifically designed to produce symbols, diagrams and other graphics automatically drawn by a computer. The information may be transmitted to remote locations instantly or during a number of seconds, depending on the amount of detail. The graphics are shown on a TV monitor that is a part of a video display terminal. Although these systems can be used in many different ways, their graphics capability has made them especially attractive to industry, business, and scientific research, where they perform many design and drafting operations. They are used, for example, to construct parts drawings, logic diagrams, schematics, architectural designs, and maps. In some cases, a computer graphics system is interfaced to a host computer to provide a sophisticated data processing and production management system. The graphics information may be stored on tapes, magnetic discs or in computer memory; printers and plotters reproduce the drawings on paper.

Televideo systems transmit not only graphics but also pictures--in short, anything that can be captured by a video camera. The picture is transmitted during a number of seconds and shown as a still, or "frozen" image on TV monitors at remote sites. These systems, usually referred to as slow scan televideo, have primarily been used for security and surveillance, signature verifications, and other banking and industrial functions. However, because of their greatly improved performance, they are increasingly being used in video teleconferencing in education, business, health and other areas. Audio and visual information can be stored on regular audio tape, or a video frame-grabber

can be used to reproduce the image on paper.

Facsimile, like any paper-copying equipment, reproduces information on paper. However, there is one important difference--the copies of the original document are made at distant locations. Facsimile equipment reproduces many kinds of information--typewritten pages, graphics, pictures. The transmission time is delayed, typically ranging from two to six minutes, although sub-minute machines are now available. Facsimile machines are widely used in business and industry.

The creative use of any of the systems can provide a variety of visual information. The unit may be quite simple and inexpensive, such as an electro-mechanical pen, and yet transmit many kinds of hand-drawn graphics. It may also be highly sophisticated and combine units for even greater versatility. For example, a graphics tablet or video writer could be interfaced to a slow scan televideo system to transmit instant graphics, still pictures, or graphics superimposed over a background picture. The design could also include the ability to recall a previous frame of information or to alternately switch between a picture and graphics. With imagination (and the technical ability to interface the equipment properly), a system could be designed to meet almost any visual need.

Almost all of these devices are compatible with both regular dial-up telephone lines and dedicated teleconferencing networks, using a standard RS-232-C interface. The exceptions are those systems that transmit high amounts of information, such as a color slow scan televideo, which requires a special modem, and highly sophisticated computer graphics models which operate most effectively on data-grade telephone lines. Some facsimile machines are limited to point-to-point transmission.

With all of these systems, audio and visual information can be transmitted simultaneously on a four-wire network. However, because the graphic information usually takes up the entire telephone bandwidth, voice and graphics must be transmitted sequentially, or alternately, on a two-wire line.

The Problem of Limited Bandwidth

Although there have been rapid advancements in narrowband visual systems, none of the devices currently available have overcome the problems of limited bandwidth. This restriction has resulted in compromises in graphics capability, picture quality, and transmission time.

The bandwidth problem is due to the differences between audio and visual information. Both are analog signals, but their frequencies differ. (Frequency is measured in cycles per second and expressed as hertz.) A telephone channel is designed to transmit only the frequencies needed to make speech intelligible. This frequency range is from 300 to 3400 hertz (Hz). The telephone thus has a narrow bandwidth of 3100 Hz, or 3.1 kHz. FM radio, which has a bandwidth of 15 kHz, is also a narrowband channel. Visual signals, however, have a much wider range of frequencies. Broadcast TV or coaxial cable have a bandwidth of about 3.5 megahertz (MHz). One wideband channel is thus roughly equivalent to 1000 voice-grade telephone lines. On a wide band channel, a full video picture containing over 2 million bits of information can be transmitted in 1/30 second. It would take approximately 10 minutes to transmit the same amount of information over a voice-grade line, if the data were not manipulated or compressed in some way (Meyer, 1976).

How, then, is it possible to transmit visual information over a narrowband channel given the limited bandwidth? Most graphic material does not contain nearly as much information as a picture. In addition, it is either black or white and has no gray tones. Handwriting, simple diagrams, charts, and graphics can thus be transmitted in real-time. However, complex graphics and pictures contain more information than can be instantly transmitted. The information can be sent on a narrowband channel, but it takes longer to reach its destination. No one wants to wait 10 minutes to receive a picture over a telephone line. Given the narrow bandwidth, the transmission time can be decreased only by reducing the amount of information. Any picture contains redundant elements. This is especially true of animated video in which there is not really much change from frame to frame. If this redundancy is eliminated, the total amount of information is reduced and the picture can be transmitted faster. There is, however, only so much information that can be omitted before the image becomes distorted. For a telephone channel, this threshold is approached before the transmission time can be reduced to zero. For example, a slow scan televideo system eliminates animation, reduces the resolution and gray shading, but it still takes a minimum of 8 to 78 seconds to transmit the picture.

The bandwidth problem is the subject of much research, and many techniques are being developed to decrease transmission time without overly affecting the quality of the picture. These techniques for data or bandwidth compression use various mathematical transformations and coding methods to more efficiently eliminate redundant information. As [illegible] electronic devices become available to perform complex coding operations, data compression techniques will become increasingly more effective in reducing transmission time.

Evaluational Considerations

When evaluating any of the telewriting, televideo, or facsimile devices, there are a number of factors that should be considered in addition to the kinds of information each one can transmit.

1. Image quality
   a. Resolution
   b. Gray scale or color scale
2. Image storage capacity
3. Transmission time
4. Cost
5. Analog or digital signal processing
6. Compatibility with auxiliary units
7. Adaptability to future technical improvements
8. Ease of operation
9. Reliability

Of these considerations, image quality, storage capacity, transmission time, and cost are so closely interrelated that an improvement in one usually has adverse affects on the others. If the quality of the image is improved, the picture takes longer to transmit. A better quality picture also requires more storage capacity, increasing the cost of the system. Given marketing constraints, manufacturers often opt for a trade-off among these factors, primarily balancing cost and performance. This interrelationship may be better understood if each factor is described briefly.

Image Quality

The quality of the graphic or pictorial information generated by a visual system depends on its resolution and, in some cases, its gray or color scale.

Resolution refers to the total number of points, or individual elements, that comprise an image. The greater the number of elements, the higher the resolution. In some systems, such as electro-mechanical pens, graphics tablets, and facsimile, resolution is expressed as lines per inch. In systems that show the image on a TV monitor, resolution is usually expressed as the total number of lines and elements in the picture. This can be most easily visualized as a matrix of dots, as shown in Figure 1.

Higher Resolution — Lower Resolution

Figure 1. A Matrix of Picture Elements

Each dot represents picture information; therefore, the greater the number of dots, the better the image quality. For example, a standard TV picture has 480 horizontal lines composed of 640 elements. If the resolution is reduced to, say, 175 lines with 75 elements, there is much less information and the picture will be distorted and of poor quality.

In many images--most graphics and facsimile--there is no shading. An element is either black or white. Half-tone pictures require variations in gray levels. Thus the quality of the image also depends on the number of gray shades representing the original picture. In slow scan televideo, for example, the gray scale varies from 16 to 64 shades. A picture with 16 shades shows a contouring effect with noticeable jumps between gray tones. With 64 steps, the shading appears continuous, much like a photograph.

It is difficult to state what degree of resolution and gray shading is acceptable. It depends, of course, on the amount of detail in a picture. However, there is another factor which is the subject of many research studies--the degree to which humans do not perceive discontinuities but rather automatically supply the missing information (see Sharma and Netravali, 1977; Sakrison, 1977). In this sense, a TV picture with 300 lines may not appear to be that much different than one having 480 lines, especially when viewed at a distance.

Storage Capacity

Except for electro-mechanical pens and facsimile, all of the systems show the graphics and pictures on a TV monitor. The transmitted information, whether it is in analog or digital format, must therefore be converted to composite video signals. Most systems are designed to interface with a CRT (cathode ray tube) monitor that must be refreshed by a separate storage unit that holds the picture information. This information is rapidly and constantly scanned to produce the video signals.

The storage unit may be an analog storage tube, magnetic disc, or digital memory. Of these devices, a digital memory has distinct advantages. If a storage tube is used, the picture will decay in time. A digital memory constantly refreshes the picture as long as the power is on. It also has a long life and needs little or no maintenance, which is not usually the case with a magnetic disc. However, a digital memory may mean a lower quality picture.

The quality of the TV picture depends on the capacity of the storage unit. The monitor may be capable of displaying a high resolution picture with 480 x 640 elements, but the storage unit must be able to hold this amount of information. A digital memory often compromises on the quality of the image because of the great number of bits needed to represent a picture with this resolution. The primary factor is cost. A picture that is twice as good in quality requires at least four times as much memory storage at a proportional increase in cost.

Transmission Time

The time it takes to transmit a given amount of information depends on the capacity of the channel. Capacity refers to the rate at which information can be sent without error, and it is expressed as bits per second (bps). Although a channel's capacity is theoretically proportional to its bandwidth, it is usually less due to noise in the channel and other factors that cause errors. A regular telephone channel has a capacity to transmit 600 to 9600 bps, depending on the grade of the line, type of modem used, and signal-to-noise ratio. An ordinary line usually transmits 1200 to 2400 bps. Speeds of 4800 and 9600 bps are possible, but they require conditioned lines and special modems, which are more expensive.

Although the limitations of a narrowband channel often result in the delayed transmission of detailed graphics and pictures some devices transmit information faster than others. For example, slow scan televideo systems vary from 8.5 to 78 seconds or longer; facsimile ranges from 35 seconds to 6 minutes. A faster transmission time often means a lower quality picture because less information is transmitted. However, it may also be due to the application of more sophisticated technology. For example, one slow scan system uses a data compression technique called differential pulse code modulation to reduce transmission time while preserving picture quality. It requires a special modem, however; it is also the most expensive of the slow scan units.

Analog and Digital Signal Processing

Most of the telewriting and televideo systems convert the graphics and pictures, which originate as analog signals, into a digital format.

A digital format has two major advantages in transmitting visual information. First, digital signals can be manipulated more readily than analog signals. This consideration is especially important if different coding schemes or data compression techniques are to be used. Second, digital pulses are regenerated as new, clean pulses at each repeater station so they are not progressively distorted by channel noise. In analog transmission, the signals are amplified at each station and the noise, which is also amplified, accumulates during the transmission time (Martin, 1976).

The conversion process is performed by an electronic component appropriately called an analog-digital converter. In brief, the amplitudes of the analog signals are quantized into discrete set of values, as shown in Figure 2. These values are then sampled at specific points, and each sample is coded to produce a stream of binary (0,1) bits. The pulse stream which carries the video or graphic information, is modulated (usually

by frequency shift keying) to a carrier wave for transmission over the telephone line.

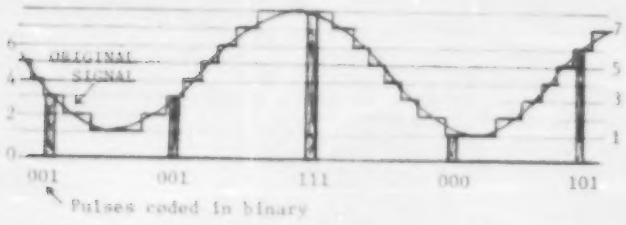

Figure 2. An Analog Signal in a Digital Format

For both narrowband and wideband channels, there is a strong trend toward coding the analog signals and transmitting them in a digital format. The telephone itself is beginning to convert to all-electronic networks of solid-state devices and computer control that are ushering in a new era of electronic mail, programmable telephones, and the digital transmission of voice, video, and data. (Erikson, 1975, 1977; Gundlach, 1975, 1977; Cole, 1975; Smith, 1977).

Telewriting Systems

Telewriting systems range from relatively simple electro-mechanical pens through complex computer graphics systems. Their common feature is the ability to generate graphic or written information. The material may be hand drawn using electronic pens, tablets, blackboards, and video writers; or it may be computer drawn with a computer graphics system. In most cases the information is instantly received at distant locations. The exception is detailed diagrams or schematics that may take a number of seconds to transmit.

Although telewriting systems differ in specific design and sophistication, they are based on the same principle: visual information can be transformed into voltage frequency signals by writing on a special conductive surface. Given a defined area, the exact location of each signal can be determined by measuring its vertical and horizontal position. For example, as one writes on the conductive surface of a graphic tablet with an electronic pen, voltage or frequency signals are generated. The positions of these signals are detected along x and y coordinates. Each letter or line is thus actually a series of points. After some processing, this information can be transmitted to a receiver that reproduces the writing. The receiver may be a mechanical pen that moves in response to the incoming signals, but in most cases the graphics are shown on a TV monitor.

Electro-Mechanical Pens

Electro-mechanical pens are probably the oldest of the telewriting systems. The first model, the Electrowriter made by Victor Comptometer Corporation, entered the market of telephone technologies in the early 1960's. Since then many improvements have been made in these devices to increase their performance. For example, solid-state integrated circuitry is now used for better fidelity and reliability. Their relatively low cost, direct hard-copy reproduction of writing and graphics, and ease of operation attracts many users.

Free-hand graphics and words are produced by writing on a paper-covered platen with a special pen that generates frequency signals. The signals are processed and transmitted simultaneously to any number of locations, where the information is reproduced on paper or acetate film by a mechanical pen on the receive unit. The paper is usually roll-fed and advanced automatically as one writes. Pre-printed forms are often used for routine communications such as sales requisitions and pharmaceutical prescriptions. The units are portable and can either be coupled to a standard telephone or wired for direct access to a teleconferencing network.

Examples of electro-mechanical pen devices are the Telenote/Telescreen system manufactured by Talos Systems, Inc., Telautograph Corporation's Telescriber and DY-II Telewriter, and the Electrowriter 2000 made by Infolink Corporation. Their characteristics are summarized in Table 2.

The Telenote transmitter consists of a writing area (4-1/4" x 6-1/2"), master printed circuit board, x-y plotter, and an electronic pen with no mechanical linkages. As one writes on the paper, the pen generates radio frequency signals that are detected by the plotter. The transmitter may also have an optional built-in conference speaker and microphone. The Telescreen receiver has a pen mechanism that etches the writing onto acetate film. The transparency is then projected onto a screen so it can be viewed easily by a number of people.

Telautograph's Telescriber and DY-II Telewriter models are basically the same in design and operation. The solid-state transmitter has a pen attached to a mechanical arm (servo-mechanism) that generates voltage frequency signals and positioning information. The paper-covered writing area is 3-1/2" x 5". The standard receive unit, like the transmitter has a mechanical pen with a ballpoint tip that reproduces the writing onto a 3-1/2" x 5" paper surface. The company also has a receiver with a thermal image pen for inkless reproductions. Telautograph will supply, on special order, a receiver that has an overhead projector so the information can be shown on a screen.

The Electrowriter 2000 made by Infolink operates in much the same manner as the Telautograph units. The Electrowriter is available as a transceiver, transmitter, or receiver. All have electro-mechanical pens and writing areas of 3-1/2" x 5", as shown in Figure 3. An optional device automatically prints the date and time that each message is received. Infolink is a subsidiary of Victor Comptometer Corporation.

Electronic Blackboard

The Electronic Blackboard designed by Bell Laboratories is another system that can be used to generate hand-drawn graphics and writing. What appears to be an ordinary blackboard is actually an electronic device that transmits written and drawn material over regular telephone lines. As one writes on the pressure-sensitive blackboard with a piece of chalk, the information is simultaneously shown on television monitors at remote locations.

TABLE 2

CHARACTERISTICS OF ELECTRO-MECHANICAL PENS

	Telenote-Telescreen	Telescriber/DY-II Telewriter	Electrowriter 2000
Manufacturer	Talos Systems, Inc.	Telautograph Corporation	Infolink Corporation
Transmitter	Paper platen Writing area 4-1/4"x6-3/4" Solid-state x-y plotter Electronic pen	Paper platen Writing area 3-1/2"x5" Solid-state Electro-mechanical pen	Paper platen Writing area 3-1/2"x5" Solid-state Electro-mechanical pen
Receiver	Acetate film Writing area 4-1/4"x6-3/4" Etching pen	Paper platen Writing area 3-1/2"x5" Electro-mechanical ink pen or thermal inkless pen	Paper platen Writing area 3-1/2"x5" Electro-Mechanical ink pen
Resolution	200 lines/inch		
Accuracy	1/64"		
Cost	$1095 - transmitter $1895 - transmitter with voice capacity $3095 - receiver	$1475 - transmitter $1475 - receiver	$1400 - transmitter $1425 - receiver

Figure 3. Electro-Mechanical Pen with Paper-Covered Platen

In addition to the blackboard itself, the basic system includes a control unit, a transmitter, and receiver (or two transceivers) to process the signals; 202-type modems, and a memory unit to store the information and refresh the video display. The system is compatible with almost any standard television monitor, although a 23-inch screen is recommended. The arrangement of these components is shown in Figure 4.

The blackboard is composed of two electrically conductive sheets separated by a small gap. When the flexible outer surface is touched with a piece of chalk, it contacts the inner sheet. The resulting voltages are detected and multiplexed to represent the x and y coordinates of the chalk's position. The voltage signals are sampled and digitally encoded by the transceiver and then transmitted over the telephone line at a rate of 1300 bits per second using frequency shift keying and a 202 modem. At the receive location, the signals are decoded and stored in a memory unit. The memory is scanned at the standard TV rate, and the resulting composite video signals are shown on the monitor. Because the memory is a scan-converter storage tube unit, it stores a frame of information for approximately 45 minutes. If a picture is continuously held on the monitor for a longer period of time, it begins to decay. The blackboard has a writing area measuring 42" x 56" and a resolution of 1000 x 1000 points, or about 20 points per inch. The resolution of the image shown on the monitor is about 300 x 300 points, which is comparable visually to a

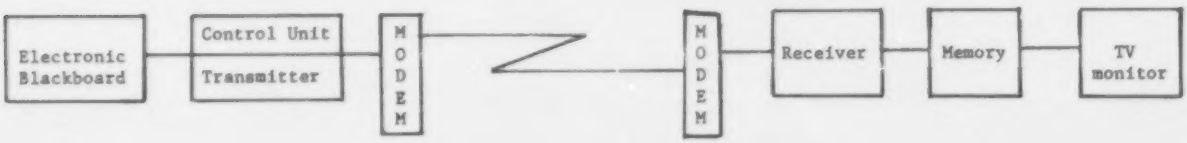

Figure 4. Basic Components of Electronic Blackboard System

standard television picture.

The control unit provides both erase and clear functions. A small part of the graphics shown on the monitor can be electrically erased by touching the blackboard at that particular point, or the entire display can be cleared.

An interactive graphics system is possible by installing transmit and receive units at each location. However, because a half-duplex channel is used, graphics cannot be transmitted in both directions simultaneously. Each location may interface up to three blackboards and three monitors to the transceiver. Voice communication is on a separate telephone line.

The hand-drawn graphics can be recorded on regular audio tape at the transmit site, or a video copier can be attached to the monitor for hard copy reproductions of the displayed images. Graphics and sound can both be recorded on a stereo recorder.

The Electronic Blackboard has so far been used only on an experimental basis. It has been tested at the University of Illinois, Urbana-Champaign, since 1974, where it has been installed at five locations to provide continuing education to engineers. Bell Laboratories is planning to file for tariffs and market the blackboard within the near future.

Light Pen-Video Writers

Hand drawn material can also be generated by writing directly onto a TV monitor with a light pen. In this case, the monitor may be thought of as an electronic grid that detects the pen's positions as it is moved across the screen. The signals representing the graphic information are then processed and transmitted to remote monitors.

An example of this type of device is the Video Writer manufactured by For-A Corporation of America. It is a self-contained unit consisting of a 12-inch monitor, semiconductor memory, and a light pen with a built-in microprocessor (see Figure 5). The TV picture area is divided electronically into more than 75,000 points, which change state when exposed to the signal emitted by the light pen. Thus, as the pen is moved across the screen, the points are interconnected to form letters and lines. The graphic information is stored in a semiconductor memory and can be distributed to other equipment connected into the system.

The Video Writer can perform a number of different functions that are controlled by the pen's microprocessor and a push-button operating panel. A straight line, for example, can be drawn automatically in either a vertical or horizontal direction by touching the pen to the line's starting point and depressing a button until the line is completed. Other functions allow partial or full erase of the graphics, up or down white-out to mask the picture, and selection of two line thicknesses.

The Video Writer is rather unique in that it is designed specifically to produce free-hand graphics for video systems. However, it is a peripheral unit and must be interfaced with other transmitting/receiving equipment. It is not clear if the graphics could be transmitted in real time on a narrowband channel, given the number of points to be coded. If this is the case, it might be feasible to implement a relatively simple data coding method to transmit only the changing points rather than an entire picture. This could make instantaneous transmission possible.

Because the Video Writer is compatible with nearly all television systems, it could be interfaced with a slow scan televideo unit. This combination of devices would provide hand-drawn graphics, pictures, and graphics superimposed over pictures. The white-out mode could be used to partially or completely mask over the background image while drawing with the pen.

The list price of the Video Writer, including the 12-inch monitor, is $5,386.

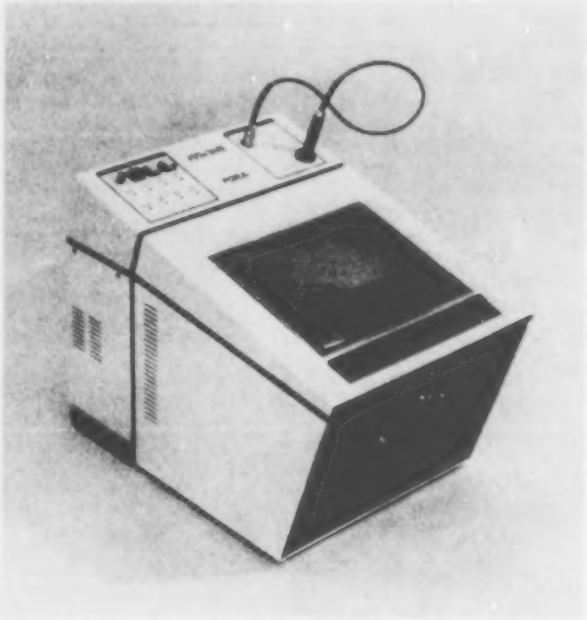

Figure 5. For-A Video Writer

A light pen is a rather simple device that contains a photosensitive cell and small-aperature lens. It can be used in a variety of ways--drawing, erasing, editing, entering data, determining coordinates. The functions it performs are controlled by a processor (microprocessor, minicomputer, or host computer) and software.

Light pens are most commonly used to determine coordinates on a CRT monitor. When the pen is placed on the screen, it responds to the brightness of the display and generates an electronic pulse that is transmitted to the computer. This interrupts the computer, which then determines the exact coordinates of the pen's position on the screen. Using the appropriate software, one can then enter data, erase or perform other functions at that point on the display. For example, the newspaper industry edits copy by touching a light pen to a misspelled word. The correct spelling is then entered on a keyboard. Industry uses light pens to assemble parts diagrams. When the pen is touched to a specific point, a sub-part called out of memory is automatically drawn at that location. Because a light pen responds to changing light, as in a CRT refresh display, it cannot be used with a storage-tube monitor.

Light pens alone can be purchased from a number of companies, including Information Control Corporation and Aydin Controls. However, they must be used in conjunction with a hardware-software package system appropriate to the user's needs. Many firms specialize in developing these systems, such as Digital Equipment Corporation and System Development Corporation.

Graphics Tablets

Graphics tablets are another type of telewriting system that can be used for graphic information. Using a special conductive pen (or stylus), graphics are produced by writing on a piece of paper that is placed directly on the tablet's surface. The tablet is an electronic device that senses the pen's positions. The graphic information is digitized and transmitted to a remote TV monitor for display.

Some tablets are designed for hand-drawn graphics only. The Audiograph system, produced by Interand C Corporation, falls into this category. However, most tablets can be used in two ways: to produce hand-drawn graphics and computer graphics. Tablets made by Tektronix, Summagraphics, and Talos, for examples, can be used with a computer graphics system, where they perform other functions in addition to the free-hand writing and drawing. All of these tablets are summarized in Table 3.

TABLE 3

GRAPHICS TABLETS

Device	Transmitter	Receiver	Resolution (inches)	Accuracy	Cost
Audiografix System Interand Corp.	Oxide coated glass Conductive pen Solid-state memory	Digital memory (standard monitor)	.01	1%	$4000 (transmitter and Tablet)
4953 Tablet Tektronix, Inc.	11" x 11" surface Digitizing grid Conductive pen Control unit	Graphics display terminal with CRT monitor	.01	±0.03	$3195
4954 Tablet Tektronix, Inc.	40" x 30" surface Digitizing grid Conductive pen Control unit	Graphics display terminal with CRT monitor	.01	±0.10 over entire tablet	$5295
4956 Tablet Tektronix, Inc.	20" x 20" and 38" x 48" surface Digitizing grid Conductive pen Control unit	Graphics display terminal with CRT monitor	.005	±0.004	$4900 and $7500
Intelligent Digitizer Summagraphics	From 11" x 11" to 42" x 60" surface Self-calibrating grid Microprocessor Conductive pen Controller	Graphics display terminal with CRT monitor	.005	±0.004	$2400 to $6000
Cybergraphic Tablet Talos Systems, Inc.	From 11" x 11" to 44" x 60" surface Electronic closed loop servo system Conductive pen Control unit	Graphics display terminal with CRT monitor	.01 (.005 optional)	±0.01	$2500 to $6045

The Audiografix system is designed specifically to transmit free-hand or traced graphics over a voice-grade telephone line or radio subcarrier. It consists of a graphics tablet, transceiver with a solid-state memory, and a conductive pen with a ball-point tip. As shown in Figure 6, a home monitor allows the user to watch the development of the graphic presentation. The tablet is actually a transparent piece of glass with a resistive oxide coating. It can be used in three different ways; the tablet can be placed directly over a standard TV monitor, it may be placed over a piece of paper for tracing graphic information, or a piece of paper can be placed over the tablet and one can write directly on the paper.

As one writes on the tablet (or paper) with the pen, voltage signals are generated and multiplexed for the x and y coordinates of the pen's position. The analog voltage signals are digitized and stored in the memory, which has a storage capacity of 1000 points in x direction and 512 points in y. The memory is then raster scanned to produce composite video signals to be shown on the home monitor. To display the graphics on TV monitors at remote locations, the information is transmitted as tone signals in an analog FM format.

The tablets made by Tektronix, Summagraphics, and Talos are similar in a number of ways. They can all be used to produce free-hand material, to digitize traced information, and to position graphics that are drawn by a computer.

For free-hand graphics only, these tablets would operate in much the same manner as the Audiographix system. One would simply write or draw on the tablet, and this information could be displayed on remote television monitors. The receiver would require memory storage capacity to refresh the video display and processing circuitry to convert the incoming digital pulses to composite video signals.

97

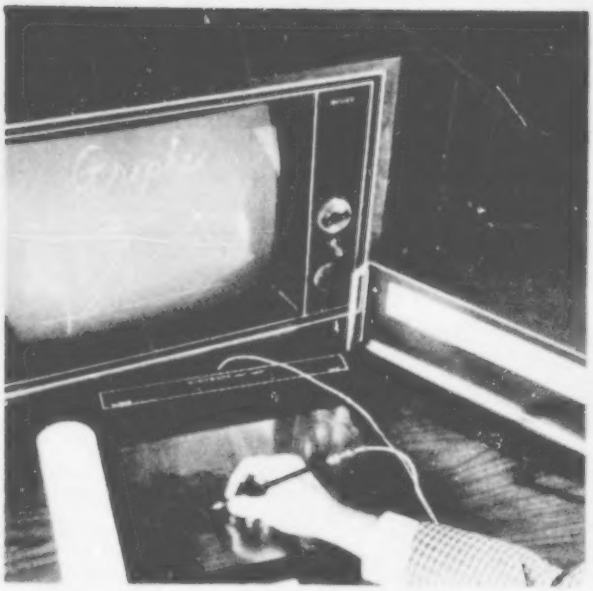

Figure 6. Interand's Audiografix System

When interfaced with a computer graphics system, these tablets perform additional digitizing and positioning functions. They can still be used for free-hand graphics, but they are also capable of digitizing detailed schematics and diagrams. In this case, prepared graphics are placed on the tablet and traced with the pen or a cursor. The information is digitized by the tablet and either shown on a TV monitor or entered into computer storage. As a positioning tool, the pen and tablet are used to determine the exact x-y locations for the insertion of computer drawn graphics--symbols, vectors, sub-diagrams. (Computer graphics systems are discussed in the next section.)

This type of graphics tablet typically contains parallel wires spaced one hundredth of an inch apart which run in both x and y directions to form a grid or matrix. As one writes on the tablet's surface, the pen and wires interact electronically to generate pulses representing the pen's position. The information is digitized as one writes. The spacing of the wires determines the tablet's resolution, which is usually expressed as a decimal number. Tablets with a resolution of .01 are capable of digitizing 100 lines per inch; those with a resolution of .005 reproduce 200 lines per inch. A typical digitizing rate is 100 coordinate pairs per second (Machover, 1977). This allows one to write at a speed of about 1 inch per second on a tablet with a .01 resolution, although speed also depends on the total bit output.

The surface area of the tablets ranges from 11" x 11" to 44" x 60". They are about 1/2" thick. The pens have either a ball-point tip or inkless stylus. For more exacting accuracy, push-button cursors can be purchased for $235 to $400.

The graphics tablets made by Tektronix, Inc., have electronic grids of iron-cobalt wires to digitize the graphics drawn on the tablet. Of the three Tektronix tablets, the 4953 is the simplest and least expensive. It converts graphics into a digital format in either a single-point or multiple-point entry mode using either the pen or an optional cursor ($235). The 4956 tablet, shown in Figure 7, digitizes graphics in four different modes: point--single-point; stream--continuous points; switch stream--continuous points only when the pen is down; and incremental--points updated only when the value changes. The 4953/54 tablets are compatible with Tektronix 4010 terminals, which cost $4450 and more, or the more sophisticated 4081 Interactive Graphics System. It costs $27,000. The 4956 tablet interfaces with the 4051 terminal, which has an 11-inch monitor and costs $7500. The graphic information can be stored on a peripheral disc or a digital cartridge tape recorder, and dry-process copies can be made on a Tektronix Hardcopy Unit that sells for $4295.

Summagraphic's Intelligent Digitizer has a built-in microprocessor that gives it greater flexibility. The microprocessor serves to implement such features as floating origin, incremental mode, binary to BCD conversion, output data formating as well as calculations of volumes, areas, linear displacements and perimeters. The tablet operates on a patented principle Summagraphics calls Magnetostrictive Ranging. It is compatible with terminals made by other companies, such as the 4010 Tektronix line.

The Cybergraphics tablet made by Talos uses a closed-loop electronic servo system to detect pen displacement. According to Talos, it is "...literally impossible to move the pen without detection." The tablet operates in five digitizing modes: single point conversion, continuous track, continuous run, remote from externally supplied signal, and incremental.

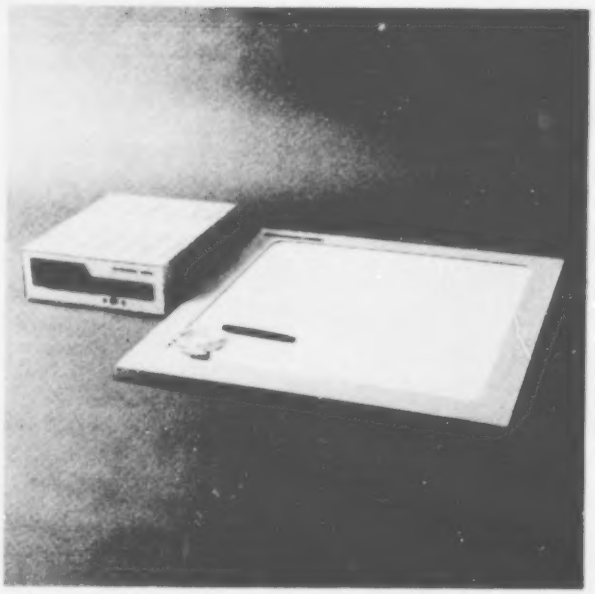

Figure 7. Teletronix 4956 Graphic Tablet

Computer Graphics Systems

Computer graphics sytems are capable of producing many types of graphics in addition to hand-drawn material. The design may be quite simple, generating lines, symbols, and simple diagrams; or it may be highly sophisticated and perform a variety of computer-controlled functions. The graphics capabilities of these systems may be summarized as follows:

1. Free-hand graphics: line drawings, graphs, writing produced on a graphics tablet or with a light pen.
2. Traced graphics: maps, schematics, structural and parts diagrams traced on and digitized by a graphics tablet.
3. Graphic symbols: programmed symbols stored in memory, selected from a menu or function keyboard and positioned by pen.
4. Computer-drawn graphics: software controlled graphics automatically drawn by a computer; position of drawing determined by pointing light pen or tablet pen at that point.
5. Alphanumerics: letters and numbers entered on a keyboard.
6. Special graphics functions: programmed graphic manipulations, such as three-dimensional rotation, zoom, magnify, reduce, label, as well as the plotting of bar charts or pie charts.
7. Hard-copy graphics: information reproduced by plotters and printers.

The kinds of information that can be generated on a computer graphics system depends on the hardware and software design. the basic blocking blocks of a system include: a graphics input device, processor, controller, and display unit or terminal. However, the arrangement of these units can be varied to provide different capabilities. Figure 8 illustrates two possible configurations.

The first configuration is a one-way system designed to transmit hand-drawn and computer-drawn graphics to a remote display unit that has a CRT monitor. A graphics tablet is used to produce free-hand drawings, writing or traced material. The control-processing unit includes a digital memory, logic circuitry, a programmable microprocessor, to provide additional graphics functions--symbols and lines, formatting, selective erase, conics and scrolling, upper and lower case letters, some mathematical operations. The information is transmitted to a remote display unit, and the graphics appearing on the CRT monitor may be automatically constructed by the computer, hand drawn, or a combination of both.

The second configuration has the same basic units as the first, but the design provides a much different graphics capability. In this arrangement, the display terminal and midicomputer interact as a sophisticated drafting system like that typically used in industry. A graphics tablet is again used to enter information, but here the graphics are more likely to be sub-part diagrams or schematics that are either shown directly on the monitor or sent to computer storage. A host midicomputer, which is time-shared by a number of customers, stores information in memory or on magnetic discs and tapes. The computer is programmed not only to draw lines and many types of symbols, but also to plot bar charts, rotate a diagram in three-dimensions, zoom, magnify, reduce, label, and perform other specialized functions. The display terminal is the drafting tool. It has a CRT monitor, keyboard, digital memory, and other circuitry. The operator sits at the terminal and begins a diagram by using the graphics tablet to designate the exact location where a symbol or sub-part is to be drawn. Operating the keyboard, the appropriate graphic information is called up and automatically inserted at that point. The drawing may also be manipulated by rotating it for a three-dimensional view, magnifying one part while reducing another, or deleting sections. The development of the diagram is monitored on the CRT. After a series of graphic oeprations, the result may be a mechanical drawing of a tractor or an electrical wiring schematic, which can be reproduced by a plotter for permanent copy.

The examples in Figure 8 represent arrangements that are virtually opposite in graphics capability. Because a computer graphics sytem is modular, it can be designed to meet many other types of informational needs. Perhaps the best way to illustrate the variety of systems possible is to discuss each of the units, or building blocks, in more detail. Machover (1977) and Lapidus (1976) discuss many of these considerations.

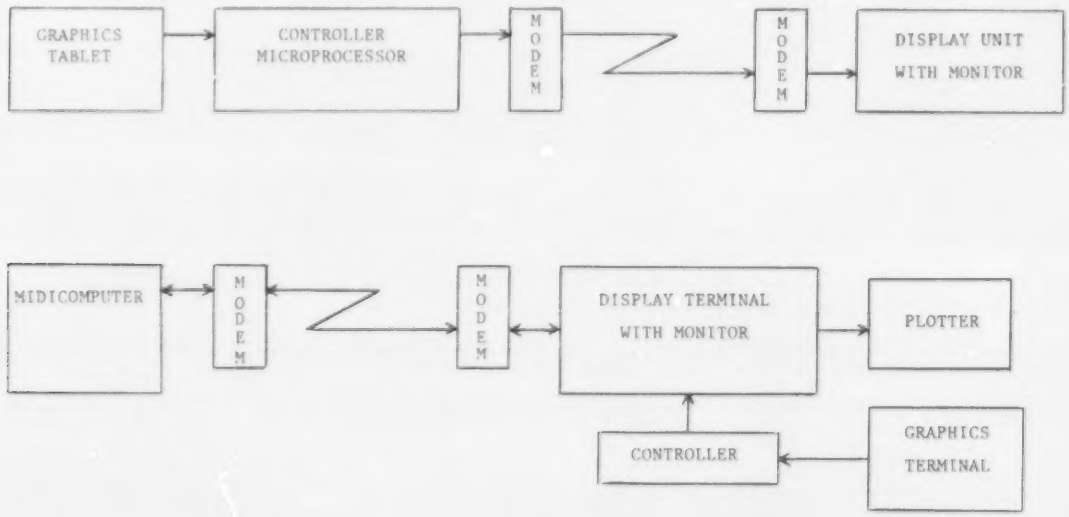

Figure 8. Two Possible Computer Graphics Systems

Graphics Processor: The graphics processor is the heart of the system. It is the computer hardware and software that provide the stored graphic symbols, interpret the operator's commands, and perform the appropriate graphics function (Coenen, 1976). It may be a microprocessor, a microcomputer, minicomputer, midicomputer, or a large main-frame computer. The graphic capability and cost of the system depend on the type of processor selected and its bit storage capacity. The larger processors have advantages in terms of greater programming flexibility, but their costs increase accordingly. Memory storage capacity can also vary widely, with 8000 bits usually a minimum amount. Many units have an add-on option so the memory can be expanded later. A microcomputer-based system can cost anywhere from $500 to $10,000. Whereas a midicomputer costs from $250,000 to $750,000, it can support a number of users in a time-shared mode.

As seen in the example configurations, the processor may be a part of the control-transmit unit or it may be a host computer. Another possibility is to design the processor into the display terminal, adding graphics functions that may or may not be supplemented by a host computer. A built-in processor is oftentimes a microprocessor. There is, however, a recent trend to increase the terminal's graphic capacity by incorporating a minicomputer or microcomputer that can be programmed to provide more sophisticated graphics.

A processing unit may also contain a graphics board, a hardware and software component that performs a number of routines. For example, a graphics board made by MiniTerm Associates interfaces to an Altair or IMSAI microcomputer. It has a programable read only memory that can generate 64 graphic characters. These are displayed in a 40 character by 20 line ASCII format with a resolution of 100 by 160 points. Some boards are also capable of producing color graphics or have higher resolutions. Vector Graphics, Inc. is another manufacturer of graphic boards. The average cost is from $200 to $300.

Graphics Input Device: A graphics input device performs two basic functions: it enters information and it positions that information in the desired location on the display. A graphics tablet is usually used for these two purposes; a light pen is another common positioning tool (see previous discussions of these two units). Information can also be entered from a keyboard, planimeter, or a microdensitometer. A keyboard is essentially used for alphanumeric data, such as text or labeling parts in diagrams. A planimeter, which has a mechanical arms, digitially generates point coordinates as it is moved over a flat surface. It is often used to measure areas on maps or engineering drawings. A microdensitometer is a type of film scanner that, like a TV camera, scans a picture to reproduce the matrix of picture elements. Graphics may also be positioned by a joystick or trackball, which define the x-y coordinates of a specific location.

Control Unit: The control unit links the input device and the processor. It is used to select the graphics and to instruct the processor to perform various functions. A graphics tablet has its own control unit and this may be sufficient for some purposes. Push-button cursors are also available to select symbols and position them on the display. In more complex systems, a control unit with a keyboard is used to instruct the processor.

Display Terminal: Display terminals vary widely in performance and design. It may be as simple as a CRT monitor and analog-to-digital converter; or it may contain a minicomputer, logic circuitry, and a keyboard to enter information and commands.

There are basically two types of CRT monitors. One requires constant refreshing and the other has its own storage capabilities. In many cases, the monitor is of the refresh type and must be updated 30 to 60 times per second by a buffer memory to maintain a constant picture. A refresh monitor has two primary advantages: it can be selectively rewritten without erasing the entire picture, and it can be used with a light pen. However, its resolution is limited by the total number of bits that can be stored in the memory. While a monitor having a storage tube with long-persistence phosphors may have a higher resolution, the total picture must be erased before a new image can be shown. One graphics system, the Tektronix 4081, differs in that it has a monitor that is both a refresh and storage type, offering the user a choice of how to best display the image.

Rather than a CRT monitor, a display terminal may have a plasma panel. Introduced on the market in the early 1970's, plasma panels are increasing in performance while decreasing in cost. The plasma display consists of two glass plates separated by a space filled with neon gas. When voltage is applied, bright orange neon dots are seen. Because the display is a direct representation of digital information, it does not require an analog-digital conversion like CRT. Panels are also twice as bright, have a wider viewing angle and less distortion (Turner, 1977; "Plasma Display Techniques," 1977). Plasma panels are capable of showing color, and pictures may be projected onto the back of the screen for front viewing.

The display's resolution, accuracy, and character repertoire are other factors that vary. One type of terminal is an alphanumeric display that has the conventional ASCII set plus standardized graphic characters. For example, a stroke generator can be used to produce 30 different line segments. These terminals typically have a resolution of 48 lines with 72 characters per line, limiting their application to simplified drawings and annotated information. However, their cost is less, generally starting at $2000.

Many display units/terminals operate in standard television formats. However, the resolution depends on the memory refresh capacity, scanning rate, and bandwidth. Depending on the unit, the resolution typically ranges from 200 x 200 elements to 480 x 640 points. Those units with higher resolutions are able to show more detailed diagrams, but they are also more expensive. High resolution monitors with 1024 x 1024 or even 4096 x 4096 addressable locations are also available for displays requiring exacting accuracy and dynamic images.

To illustrate a display terminal used for interactive graphics, the Tektronix 4010-1 model is shown in Figure 9. It is compatible with time-sharing systems, large mainframes, and minicomputers. Up to 1024 x 780 viewable points may be displayed on the storage CRT. It has a set of 63 printing characters and a generating speed of 1200 characters per second.

Communications Interface: Most computer graphics systems are designed to interface with narrowband communications channels, and the terminals usually have a standard RS-232-C communications interface. These systems typically transmit information at a rate of 2400 bits per second or less, and are thus compatible

Figure 9. An Interactive Computer Graphics Terminal

with a regular telephone channel. Speeds of 9600 bits per second or higher are possible, but special modems and data lines would be needed. The typical bit rate allows less complex graphics to be transmitted in real time, but detailed drawings and diagrams requiring a greater bit output would take longer to transmit, the time depending on the total number of bits needed to reproduce the information.

Some graphics systems are for local use only, or they require a wideband communications channel. In-house systems often have a high resolution terminal directly linked to a minicomputer. Systems requiring wideband channels are usually designed for animated and color videographics.

Products: Because computer graphics systems vary greatly in performance and cost, it is difficult to summarize products available on the market. Costs range from below one thousand to hundreds-of-thousands of dollars.

One source of computer graphics equipment is the hobby market. The Apple II microprocessor has color video graphics capability. It can display graphics in 15 colors with a resolution of 40 x 48 characters or four colors (black, white, violet, green) with a resolution of 230 x 192. The basic unit can be upgraded by adding a number of peripherals like disc storage modules. The Pet processor is a similar hobby system. Amateur radio operators are often adept at designing relatively inexpensive graphics terminals. Robertson (1973), for example, describes a unit built around an 8-bit microprocessor, 65,000 bits of memory storage, and a 12-inch monitor.

A typical interactive graphics system might range from $8,000 to $15,000. The cost also depends on the addition of peripherals, such as plotters, printers, cursors, hard copy units, digital cartridge recorders, and other devices. While some manufacturers offer a complete line of systems, individual components may be purchased from a number of companies, including just the software packages. Other firms specialize in designing systems specifically for the customer. Machover (1977) lists many of these companies according to the types of products they sell.

Many companies manufacture computer graphics systems that are primarily geared toward designing and drafting functions. These include:

Auto-trol Corporation	- AD/380 Design and Drafting System
Tektronix, Inc.	- Interactive Graphics Systems
Applicon, Inc.	- Graphics Systems
Bendix Communications Division	- Design and Drafting System
IBM-3250 Interactive Graphics Aydin Controls	- 5215 Limited Graphics System

Slightly different systems are made by Instronics, Ltd., Numonics Corporation, Optronics International, Inc., and Interand Corporation. Instronic's Gradicon Graphic system is a digital converter that employs a cursor that is moved over which digitizes prepared drawings, graphs, maps as it is moved over the surface. Numonics' Graphics Calculators are optical encoding systems that measure and digitize the areas of maps, blueprints, x-rays, as well as the lengths of river beds and highways, for example, Optronics' Digital Imaging Processors are high-speed microdensitometers, or rotating drums, that digitize, manipulate, enhance and reconstruct graphic/photographic imagery. Interand's Telestrator system is designed for wideband communications channels. It basically superimposes hand-drawn graphics and special graphic symbols over animated video pictures. Although it is designed for wideband communications channels and the transmission of animated pictures, it may be possible to modify this system for narrowband media by transmitting only the graphics in real time.

Slow Scan Televideo Systems

Slow scan televideo (SSTV) systems add another dimension to visual teleconferencing--the capacity to present pictorial information as well as graphics. Any image that can be captured by a video camera can be shown on a televideo system, including views of the instructor and classroom, outdoor scenes, handwritten and prepared graphics. The picture is transmitted over a regular telephone line or FM radio channel during a number of seconds and shown as a still image on a TV monitor. In most cases, the image forms gradually, starting at the top of the screen and moving down.

Slow scan televideo is, at this time, the only system capable of transmitting pictures over a voice-grade telephone line. As such, it has much potential in education, business, industry and other fields (Louv, 1977; Meyer, 1976; Meyer, 1977; Cohen and Meyer, 1975; Meyer, undated). The pictures are not animated, but movement may not be needed in many situations. Still images may actually be preferable, for they allow people to concentrate on the message by eliminating irrelevant motion. It may also be easier to remember a series of single pictures than continuously moving information.

The Picturephone developed by Bell Laboratories does transmit animated video. However, because it requires a special transmission channel of 1 MHZ, AT&T has virtually halted its installation and less than 1000 Picturephones are still in use. It displays 30 frames per second, which is similar to broadcast animation, and has a resolution of 251 lines with 211 picture elements per line. Bell Labs is working on a version of this unit that generates graphic data with a light pen and has a plasma panel rather than a monitor for display (Louv, 1977; Gundlach, 1975).

The basic units in an SSTV system are a video camera, transmitter, receiver (or transceiver), and TV monitors. Closed circuit television (CCTV) cameras and monitors are usually used. These units can be arranged in various combinations to provide one-way and interactive videographics between locations. Figure 10 illustrates two configurations. Multiple cameras can also feed one remote receiving site, or multiple cameras and monitors may be used at both the transmit and receive locations. In addition, auxiliary devices can be interfaced to the system. These units include video pointers, tape recorders, large-screen projectors, and frame-storers/grabbers for extra frame storage and hard copy reproductions of the images.

The core of the system is the transmitter and receiver or transceiver, which are basically scan converting devices. Scan conversion makes it possible to transmit a picture over a narrowband channel and view it on an ordinary TV monitor. In this process, the fast-scan (3.5 MHz) video signals are converted, or slowed down, to the allowable audio frequency of 3 kHz. This results in a longer transmission time that ranges from 8 to 78 seconds. The slow-scan signals are then reconverted at the receive site so the picture can be shown on a TV monitor.

The increased performance and flexibility of SSTV systems is primarily due to improvements in the scan converting units. Slow scan was developed in the late 1950's by Copthorn McDonald, but the transmitted image could not be reconverted into a conventional TV picture. Thus, a special monitor with a phosphor tube was used, and the image was of very poor quality (Louv, 1977). SSTV systems now use an analog storage tube, magnetic disc or digital memory to reconvert the signals and display the picture on a TV monitor. Of these three, a digital memory has distinct advantages. If an analog storage tube is used, the picture will decay in time. A digital memory constantly refreshes the picture as long as the power is on. A digital memory also has a long life and needs little or no maintenance, which is not the case with a magnetic disc (Steber, 1975; Ingram, 1976).

Digital signals can also be manipulated in more ways than analog, allowing, for example, different coding schemes and faster transmission times.

Selected SSTV Systems

Robot Research, Inc., Colorado Video, Inc. (CVI) and NEC-America, Inc. manufacture sophisticated SSTV systems that transmit pictures over a voice-grade telephone line. These systems are:

- Robot Phone Line Television Systems:
  - 520 transmitter
  - 510 receiver
  - 530 transceiver
- CVI Narrowband Video Systems:
  - 260B transmitter
  - 275A receiver
  - 275B receiver
- NEC Telephone Video Systems:
  - 754 transceiver
  - 751 transceiver (monochrome and color)

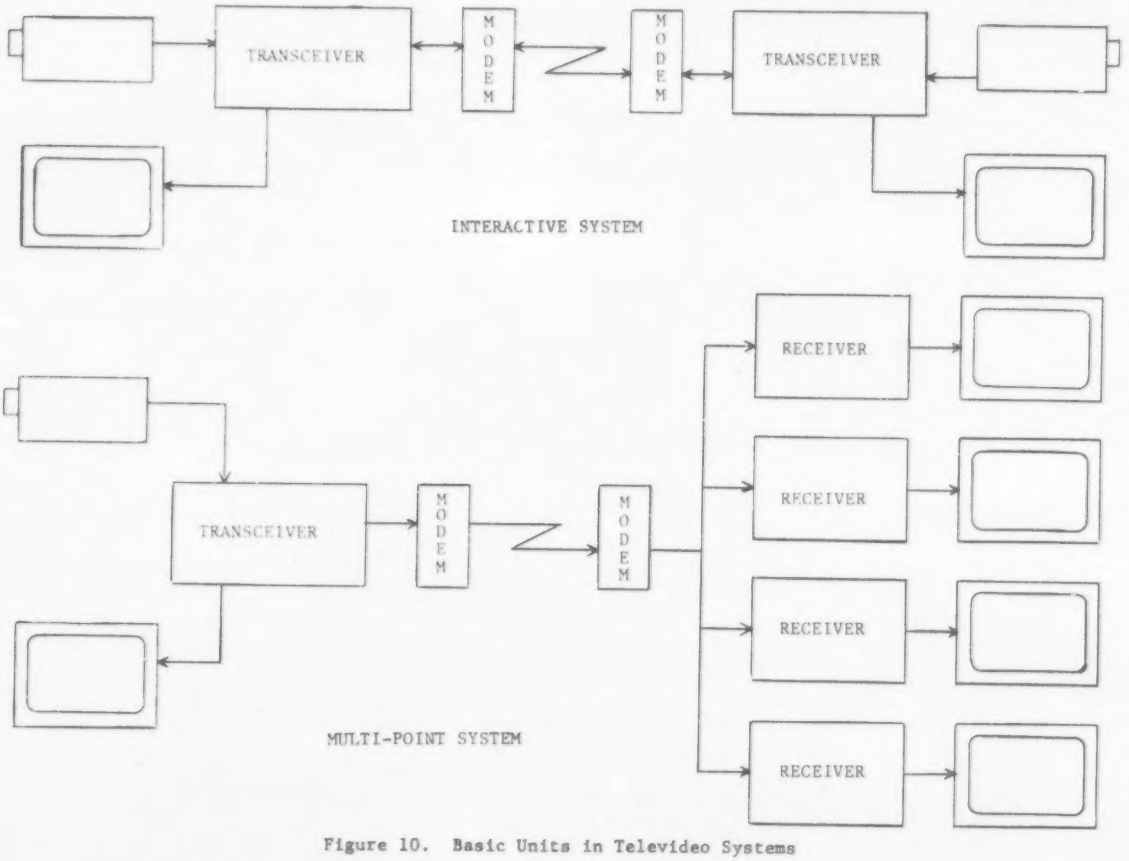

Figure 10. Basic Units in Televideo Systems

All of these units transmit monochrome pictures except for NEC's model 751, which transmits both monochrome and color. For color pictures, it requires a special modem. CVI also has a model 201A transmitter that operates on an 8 kHz bandwidth, such as an FM radio subcarrier. The transmission time is from 8 to 30 seconds, depending on the resolution and operating mode desired. The cost is $1950. It must be used with an 275-8 kHz receiver.

The Robot, CVI, and NEC systems have dynamic random access memories (RAM's) and integrated circuitry to store and process the picture information, and they all digitize the video signals. In addition, they share the following characteristics:

1. They accept standard monochrome, 525-line composite video input signals (1V peak-to-peak, 75 ohms). The NEC 751 unit accepts both monochrome and standard NTSC color signals.

2. The picture may originate from a video camera or a tape playback machine.

3. Using different camera lenses, the subject matter can range from handwriting to outdoor scenes.

4. They are compatible with a standard 525-line TV monitor (the NEC 751 can be used with an ordinary color TV set).

5. Signals are FM modulated for transmission (NEC 751 color unit uses pulse code modulation and predictive encoding to reduce the data.

6. Voice and pictures can be transmitted alternately on the same telephone line, and they can be transmitted simultaneously on a four-wire network.

7. Pictures can be transmitted automatically in a continuous mode or as single frames under manual control.

8. The TV picture is continuously refreshed and will not decay over time.

9. Video and audio signals can be recorded on conventional audio tape cartridges or cassettes for later playback (the NEC 751 color unit is compatible with any video tape recorder.

10. They can be interfaced with an auxiliary frame grabber/freezer as well as with a large-screen projector.

11. They cannot display a standard 300-word, 8-1/2" x 11" page, as this would require a very high resolution of 1000 lines.

Digital Scan Conversion: In addition to scan conversion, a digital memory stores the picture information and digitizes the signals. To perform these functions, it is composed of a core of memory cells, analog-to-digital converter, slow and fast scan clocks, sync circuits, digital-to-analog converter, and a series of input and output line buffers to clock the video in and out of the main memory. A typical arrangement of these components is shown in Figure 11.

The scan converter provides the appropriate circuitry to speed up the slow scan signals so they can be viewed on a conventional TV monitor, which displays 525 interlaced lines at a rate of between 15,000 to 16,000 lines per second. To do this, the incoming slow scan signals are demodulated and enter an analog-to-digital converter that changes the voltages into digitized binary equivalents. For example, in the Robot system, each line of slow scan video is sampled 256 times, resulting in 256 words that contain four bits of gray-scale information. The scan conversion takes place in the line buffers, which accept each line of digitized information and add it to the memory at the appropriate time. Timing is controlled by sync pulses that trigger the clocks. The memory operates at a fast-scan rate to continuously recirculate the picture information. This information is then converted from digital bits to analog video signals that are shown on the monitor (Steber, 1975; Ingram, 1976).

Operational Features: These units also have similar operating controls which allow some flexibility while maintaining simplicity. As shown in Table 4, pictures can be transmitted automatically or as single frames under manual control. A picture may also be held on the screen and viewed for any length of time. CVI's 260B transmitter has a more sophisticated remote control panel than does Robot's. With the CVI panel, one can select three separate video input sources, start or stop picture transmission, pause in the middle of a transmission for split-screen effects, and control an auxiliary frame freezer. Robot's 520 and 530 models have a test-memory-input switch that displays an internally generated gray scale. A switch on the 530 also allows one to view either live video or the slow scan image on the station monitor at the transmit location. The NEC 751 remote control panel has buttons that can be used to rewrite only the upper or lower half of the image.

Although SSTV systems are relatively simple to operate, some assistance or training would be needed, especially to transmit a picture. At the transmit end, one has to focus the video camera, tune the brightness and contrast levels, select continuous or single frame mode, and monitor the out-going video image. At the receive end, one only has to tune the picture quality and select the receive or hold mode.

Performance: The important differences among the CVI, NEC, and Robot systems are the picture quality, image storage capacity, transmission time, and cost. As was pointed out earlier, these factors are interrelated and result in trade-offs. This association can be seen by comparing the characteristics summarized in Table 5.

Picture Quality: The quality of the picture shown on the TV monitor depends on its resolution and gray (or color) scale. As discussed previously, each line of resolution is composed of a number of "dots" or picture elements (pixels). For a monochrome picture, each pixel is assigned a brightness value ranging along a gray scale from black to white. The more steps or levels in the gray scale, the more continuous is the shading effect. The total picture content is the algebraic product of the pixels and gray levels; the greater the picture content, the better its quality.

The Robot transmitter samples the video input signals and reduces the picture to a frame of information with 128 lines with 128 elements for a total of 16,384 pixles. Each pixel is assigned 1 of 16 gray levels. Thus the picture is represented by a total of

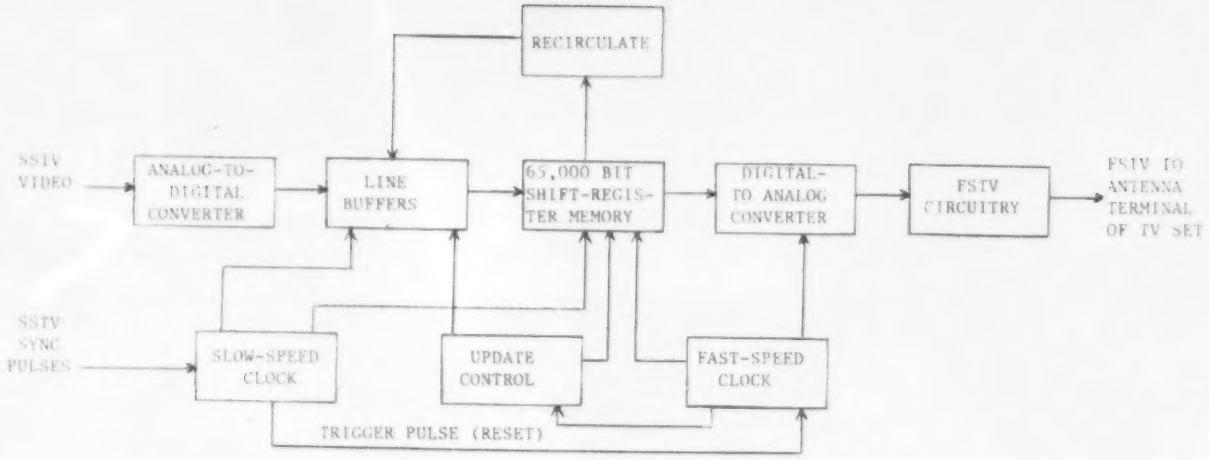

Figure 11. Typical Digital Scan Converter

65,536 bits of information (128 x 128 x 4, where 4 is the number of bits used to code 16 gray levels). This discrete array of picture elements gives the image a distinctly dotted appearance, called quantization. The 16-step gray scale also results in contouring of the individual gray shades. The quantization and contouring effects are especially noticeable when the picture is viewed at close range or on a larger monitor, for example, a 17-inch screen. Robot is developing a 256 x 256 resolution unit, but marketing plans are as yet indefinite. The Robot picture also has a square format, or 1:1 ratio. When shown on the TV monitor, the right and left sides of the screen are blanked.

The CVI 275B receiver produces an image with a resolution of 256 x 512 elements (131,072 pixels). Six bits are used to code each pixel as one of 64 gray levels, so the picture has a total of 786,432 bits of information. In comparison to Robot, the CVI picture contains more than 8 times the number of pixels and 12 times the total number of bits of information, resulting in much more picture detail. This minimizes the quantization and contouring effects, and the CVI picture is comparable to an ordinary TV image. The picture also has a 4:3 ratio.

NEC's units transmit one full picture field composed of 750,000 bits of information, on a resolution of about 262 x 340 pixels. The 754 monochrome image has 64 gray levels. The 751 model is unique, for it displays a picture in full NTSC labor.

Image Storage Capacity: The image storage capacity refers to the total amount of picture information that can be stored in the transmitter's and receiver's memory. A digital memory stores the resolution and gray scale values as bits of information in binary (0,1) form. The location of each pixel is stored as one bit; additional bits are needed for the gray shades.

The relationship between the quality of the picture and the storage capacity can be illustrated by using Robot's system as an example. The Robot picture has a resolution of 128 lines and 16 gray levels, which requires a memory storage capacity of 65,536 bits (excluding buffer memory). If the resolution is doubled from 128 to 256 lines, four times the amount of picture information has to be stored (the number of lines is roughly proportional to the square root of the number of pixels). If the gray scale is increased from 16 to 64 shades, two more bits per pixel are also required. Thus, by doubling the resolution and improving the gray scale, the memory storage requirements are increased six times, from 65,536 to 393,216 bits.

This increase in storage capacity yields a higher quality picture, but it also means a substantial increase in cost. This relationship is evident when the Robot and CVI systems are compared:

Receiver	Storage Capacity (bits)	Cost
Robot 510	65,536	$ 1,395
CVI 275A	393,216	9,000
CVI 275B	786,432	12,000

The difference in cost between the Robot 510 and CVI 275A compared to the relatively lower cost of the CVI 275B may reflect, in part, the pricing of memory chips--the cost per chip decreases after a certain number.

Transmission Time: Given the limitations of a narrowband channel, the transmission time is probably the most difficult problem in televideo systems. The best voice-grade telephone line carries 9600 bits per second, and an average line carries only 1200 or 2400 bits. How then can a complex video image be transmitted in a reasonable amount of time?

The most obvious solution is to reduce the quality of the picture. Robot samples the input video signals to decrease the picture content to about 65,000 bits. This information is transmitted in 8.5 seconds. Compared to Robot, the CVI systems have relatively slow transmission times of 39 and 78 seconds; however, they are transmitting more information and, consequently, a higher quality image. Excluding the cost factor, the difference between Robot and CVI system is essentially image quality vs transmission time.

NEC's 751 system, on the other hand, transmits 750,000 bits of information, representing one field of full NTSC color, in 75 seconds at a rate of 9600 bps. This relatively fast time is attributable to a coding scheme called differential pulse code modulation. It is used to reduce the number of bits needed to code the color values, resulting in a 2:1 compression ratio. A

TABLE 4

PHYSICAL SPECIFICATIONS OF SELECTED SSTV SYSTEMS

Transmitters	Dimensions (Approximate Inches)	Weight (Pounds)	Operating Controls		Modulated Carrier
Robot 520	12 x 11 x 7	12	Power: Off/on Video: brightness contrast Display Test memory input Audio level Frame grab: automatic, single step	Remote frame grab jack: automatic, single step	1.9 V p-p, 900 ohm 1500-2300 Hz, 1200 sync
CVI 260B	5 x 19 x 12	30?	Power off/on Video level Black level Scan: single/continuous	Remote: video source 1/2/3 Pause/scan Send Freeze Reset	1V peak-to-peak 600 ohms, 1500-2300 Hz
Receivers					
Robot 510	12 x 11 x 7	12	Power off/on Receive/Hold Width (picture scale) Contrast, brightness	Remote: Receive/Hold	
CVI 275	10 x 19 x 15	30	Power off/on Receive/Hold	Indicators: AC power scan black level white level	
Transceivers					
Robot 530*	12 x 11 x 7	12	Power off/on contrast, brightness Modes: Receive Hold Transmit Camera Test Scan: Automatic Frame grab Single step		1.9 V p-p, 900 ohms 1500-2300 Hz
NEC TVS-754 (Mono)	17 x 17 x 15 (main frame) 17 x 17 x 7 (power supply)	55 44	Power off/on	Remote: Transmit Record Ready	-20 to 0 dBm, 600 ohms
NEC TVS-751 (Color)	17 x 17 x 15 (main frame) 17 x 17 x 7 (power supply)	55 44	Power on/off	Remote: Transmit Ready Receive Full Upper Lower Sequence One-Shot	Modem interface: + (8+2)V -

*The 530 is a version of Robot's Model 400, which is designed for ham radio operators. In the 530, some of the control flexibility has been eliminated to simplify operation.

TABLE 5

CHARACTERISTICS OF SELECTED SSTV SYSTEMS

	Storage Capacity	Picture Elements (V X H)	Gray Levels	Trans Time Per Frame (Seconds)	Cost
Transmitters					
Robot 520	Digital memory 4k dynamic RAM 65,536 bits	128 x 128	16	8.5	1,545
CVI 260B	Stores each line of 1536 or 3072 bits	256 x 256 256 x 512	64	39 78	3,000 3,000
Receivers					
Robot 510	Digital memory 4k dynamic RAM 65,536 bits	128 x 128	16	8.5	1,395
CVI 275A	Digital memory 4k dynamic RAM 393,216 bits	256 x 256	64	39	9,000
CVI 275B	Digital memory 4k dynamic RAM 786,432 bits	256 x 512	64	78	12,000
Transceivers					
Robot 530	Digital memory 4k dynamic RAM 65,536 bits	128 x 128	16	8.5	1,695
NEC TVS-754	Digital memory 4k dynamic RAM 750,000 bits	262 x 340*	64	30	18,000
NEC-TVS-751 (Color)	Digital memory 4k dynamic RAM 750,000 bits	262 x 550*	Full NTSC color	75 to 300**	19,000

*NEC units transmit one field of 750,000 bits of information. Robot and CVI units transmit frames of information (two interlaced fields equal one frame).

**Transmission time depends on grade of telephone line; worst case of 2400 bits/second requires 300 seconds; 4800 bits requires 150 seconds; best case of 9600 bits/second requires 75 seconds.

special modem must be used to transmit the data at 9600 bps over a telephone line.

Other Slow Scan Equipment: Other companies that manufacture slow scan equipment include:

Hughes Industrial Products
Princeton Electronics
Alden Electronics
Venus Scientific, Inc.
Javelin Electronics
Media Concepts, Inc.
Teltron, Inc.
Infoteknics, Inc.
Hitachi Sales Corporation of America
Westinghouse
Spatial Data Systems

Hughes and Princeton Electronics both manufacture storage-tube scan converters. The Hughes model 639 has a storage capacity equivalent to 750,000 bits. This information is transmitted in 100 seconds or stored for a minimum of 25 minutes. The cost is roughly twice that of the CVI or Robot transmitters.

Alden Electronics have frame grabbers that reproduce slow scan pictures onto electrosensitive recording paper (similar to facsimile). The units interface to a scan converter. The model 600 records a frame of video in 40 seconds on a paper copy that is about six inches wide. Alden also has an 11-inch recorder that can be used to reproduce images at remote locations. Although the recorder is compatible with other SSTV systems, such as CVI, Alden prefers to interface the unit to a Hughes scan converter and an Alden slow scan

modem. The picture is transmitted in 100 seconds, and the printout has a resolution of 65 lines per inch.

Venus Scientific has a basic system for ham radio operators. Its camera-transmitter transmits a picture with a resolution of 128 x 128 picture elements and six or seven gray levels in 8.5 seconds. Its receiver uses a phosphor storage tube. Venus is planning to market a unit next year that has a digital memory with a capacity of 16 gray levels.

Javelin Electronics and Media Concepts have simple scanning systems that are used for security purposes in banks and other institutions. Each of these systems consists of a video camera, frame freezer, and video tape recorder.

Teltron makes only the storage tubes that can be used in an SSTV system rather than a digital memory.

Infoteknics has a video display system called Teledisc. Information about this system was not available from the company. According to Meyer and Brown (1976), the Teledisc system uses magnetic discs to transmit a picture over the phone lines. One magnetic disc, rotating at 1800 rpm, stores a standard TV frame. The disc is then slowed to rotate once every 84 seconds, and the TV frame is transmitted to another machine that has a disc rotating at the same speed. After 84 seconds the speeds of both discs are increased and the picture is displayed.

The Hitachi MS-200 Frame Grabber is also a magnetic disc unit. It has a rotating speed of 3600 rpm. Each video field is stored on a separate track and multiplexed to generate a complete frame. It sells for less than $1000.

Westinghouse will apparently develop and build, on special order, a two-terminal SSTV system for about $80,000.

Spatial Data Systems manufactures the EyeCom Picture Digitizer and Display, which scans and digitizes pictures for computer processing. EyeCom is primarily an in-house system used for image enhancement. For example, the slow scan picture shown on the display terminal can be graphically analyzed or written titles and supplementary data can be overlaid on the image.

RCA previously manufactured an SSTV transceiver called Video Voice (see Cohen and Meyer, 1975; and Meyer, 1976 for discussions of this system). Video Voice units are no longer built or operationally supported by RCA, but surplus models can be purchased for about $2400. Video Voice transceivers have an image storage tube and transmit a picture with 480 x 512 or 480 x 256 pixels in 55 or 30 seconds, respectively.

Facsimile Systems

With a facsimile system, documents can be transmitted over long distances and reproduced at remote locations within a matter of minutes or seconds. The material may be typewritten text, diagrams, engineering drawings, business forms, computer printouts, illustrations--virtually anything that is shown on a piece of paper.

A facsimile system consists of two machines: A transmitter and a receiver (or two transceivers). The transmitter converts the printed information into electrical signals that can be sent over an ordinary telephone line. The conversion is done by a scanner, often an optical device, that rapidly scans the page line-by-line in either a horizontal or vertical direction. The resulting electrical signals are then modulated to a carrier wave for transmission. At the receive site, the signals are demodulated and the information is reproduced on paper. In some receivers, electrosensitive recording paper is placed on a high-speed rotating drum. Each revolution of the drum corresponds to a line of print. When the writing mechanism, which is simply a piece of fine wire, touches the paper, it passes electrical current through the paper to the metal drum to "burn" the information onto the page. To operate properly, the transmitter and receiver must be carefully synchronized. A series of sync pulses are therefore sent along with the signals that represent the printed information (Heiserman, 1971).

Facsimile, like other narrowband visual systems, has been greatly improved in the last few years, especially in performance and standardization. The business facsimile market opened up in the mid-1960's, but most of the equipment was special-purpose and not standardized. By the early 1970's, machines were available that could transmit a page of information in six or four minutes. Three-minute and two-minute models soon followed. With the application of microprocessors and solid-state electronics, facsimile equipment is changing rapidly. New sub-minute machines were introduced in 1977; and slower equipment has increased in performance, offering a variety of automated features. While there was previously little standardization on scanning, synchronization, and transmission procedures, most of the six- and four-minute units are now generally compatible. Groups such as the International Telegraph and Telephone Consultive Committee (CCITT) are working to standardize the high-speed machines for both domestic and international use (Krallinger, 1974; "Facsimile: 1978 Overview", 1977).

Facsimile was actually invented 130 years ago and patented by Scottish Inventor Alexander Bain as an "automatic electromechanical recording telegraph." When photoelectric scanning was introduced in the early 1900's, facsimile became a more feasible method of transmitting information, and the first network was set up in Germany. AT&T, RCA, and Western Union began experimenting with facsimile in the 1920's, and AT&T's Telephoto and RCA's international radiophoto systems were soon in operation. Newspapers were early users of facsimile. The Associated Press acquired AT&T's network and renamed it Wirephoto, and other newspapers soon developed their own picture transmission networks. Today facsimile is often used to exchange information in business and industry, especially in manufacturing, communications, printing and publishing, finance and banking, transportation, utilities, and retailing and wholesale. Using communications satellites and land lines, international facsimile networks also link people worldwide ("Facsimile: 1978 Overview," 1977).

Most facsimile machines can be acoustically coupled to any standard telephone to transmit information over ordinary voice-grade lines, or they may be hand-wired to a leased line. However, the increasing popularity of these systems has led to a number of specialized facsimile networks. South Pacific Communications offers a common carrier service called Speedfax that allows a customer to send a page of information from New York to Los Angeles, for example, at a cost of about 20¢. While it accommodates regular analog machines, its dial-up digital channels are capable of sending information at a rate of 9600 bps and are thus compatible with the new sub-minute digital equipment ("Facsimile: 1978 Overview," 1977). ITT Domestic Transmission Systems recently intro-

duced its new FAX-PAC service. As a nationwide packet-switched data communications network, it provides a common link between facsimile equipment that has previously been incompatible. It utilizes 10 major computer centers and 14 concentrator sites to convert the software protocols, codes, and speeds of facsimile machines, allowing most of the commonly used equipment to interact. The network also features a store-and-forward service--the customer can store the message at a network location and forward it later to compensate for time zone differences or take advantage of lower rates. Messages can also be sent simultaneously to different locations. FAX-PAC is the first of the telecommunications services to be offered in the COM/PAK network, which in the future will link not only facsimile machines but also teleprinters, video display terminals, and processors into a common interactive network. COM/PAK has been approved by the Federal Communications Commission.

Performance Considerations

There are a number of considerations to keep in mind when evaluating facsimile equipment. These include:

1. Resolution - number of reproducible lines per inch (lpi)
2. Transmission time
3. Multi-point or point-to-point transmission
4. Automatic or manual send and receive
5. Flat bed or drum scanning
6. Reproduction process
7. Purchase or lease (including maintenance)
8. Compatibility with other facsimile systems

Like the telegraphic and televideo systems, facsimile can transmit only so much information in a certain amount of time over a narrowband channel. The result is less-than-perfect resolution, and the copy will show some distortion. This can be seen in Figure 12, which illustrates facsimile copy with a vertical and horizontal resolution of 96 lpi sent long distance in six minutes. Although the information is readily intelligible, there is some fuzziness. Resolution also varies with transmission time. If this same material had been transmitted in four minutes, the vertical resolution would be decreased to 64 lpi. The resolution of facsimile machines ranges from 63 to 200 lpi; 96 lpi is common for a six-minute mode.

Figure 12. Example of Facsimile Copy

The transmission time is another variable factor. It ranges from six minutes to 20 seconds for a standard 8-1/2" x 11" page. Many machines have multiple transmission speeds, allowing the user to select the speed most applicable to a given document (a slower time permits a higher resolution). Documents that are smaller in size, such as 8" x 5", can be sent faster. To transmit a page in 20 to 35 seconds, the new high-speed machines operate at a rate of 9600 bps, which requires a special modem. These units also employ data compression techniques to reduce the total amount of information transmitted. In most cases, Huffman coding is used, a scheme that looks at each scan line, analyzes the amount of black and white information, and decides on the most efficient way to send it ("High-Speed Fax", 1977).

Most facsimile systems are designed to interact on a point-to-point basis, and a typical arrangement is to install a transceiver at both locations. However, some transmitters can serve a limited number of receivers in a multi-point network.

While many units offer automatic send and receive operations, some machines must be loaded and unloaded by hand. Both types have distinct advantages. A manual machine is relatively inexpensive and is often sufficient for low-volume use. Automated equipment provides greater operational flexibility and is more efficient in high-volume situations. The more sophisticated automated units have the following features:

--Unattended transmission (polling): The unit automatically answers a call, transmits the copy, and disconnects the telephone at the end of the message.

--Automatic Message Loader: Allows as many as 100 documents to be loaded without operator attendance. In some cases, the documents can vary in length from 3 inches to 200 feet, lengths can be intermixed, and the paper can be of different thicknesses.

--Continuous Roll Feed: Allows nearly 200 messages to be continuously received.

--System Assurance: Indicates that the message is properly loaded and is being received at the other end of the line. The system automatically shuts down if there is a telephone, equipment, or power failure.

A document may be reproduced either by flat bed scanning or by attaching it to a rotating drum or cylinder. Drum scanning is often used in lower-cost, manually operated units; automated machines employ flat-bed scanning, where the document is scanned electronically as it moves through the machine.

Copies are usually made on one of three types of paper: Electrolytic, electrosensitive, or electrostatic. There is only one machine on the market, the Xerox 200 telecopier, that reproduces the information on ordinary paper.

Facsimile equipment can either be purchased or leased, but 97% of all units are rented. Lease plans vary--monthly rates, yearly rates, flat rates, charge per transmission, and other arrangements are possible. Maintenance contracts also differ, and in some cases, one may choose an option-to-purchase plan, deducting a percent of the rental fee from the purchase cost.

Products

Facsimile equipment is manufactured by various companies, and most firms have a line of machines that differ in performance and price. The most prominent companies are Xerox, Graphic Sciences, 3M, and Qwip, which provide over 80 percent of all machines currently in use. Some of the facsimile manufacturers are:

Xerox Corporation
Graphic Sciences (Burroughs Corporation)
3M Company
Qwip Systems (Exxon Enterprises)
Steward-Warner Datafax Corporation
Rapidfax Corporation
Telautograph Corporation
Muirhead, Inc.
Alden Electronics
Infolink Corporation

TABLE 6.

CHARACTERISTICS OF SELECTED FACSIMILE SYSTEMS

Device	Resolution (V/H)	Transmission Time (Minutes)	Features	Approx. Cost
Manual Systems				
Qwip 1200	64/96 96/96	4 6	Two speeds Rotating drum Electrosensitive copy	$1117 $38/month
Graphic Sciences DEX broadcaster	88	3	Simultaneous Send to 10 locations Rotating drum Electrosensitive copy	$7200 $170/month
Graphic Sciences DEX I, 180, 580	88	3 6	Two speeds Rotating drum Electrosensitive copy	
Graphic Sciences DEX 700	62 88	2/4 3/6	Four speeds Rotating drum Electrosensitive copy	
Xerox 400-1 telecopier	64/96 96/96	4 6	Semi-automatic load/eject Electroresistive copy	$1710 $50/month
3M 2346	64/96 96/96	2/4 3/6	Four speeds Semi-automatic load/eject Transmission interrupt Electrosensitive copy	$60/month
Two to Six Minute Automated Systems				
Xerox 200 Telecopier	77/80 96/96 64/96 96/96	2 3 4 6	Four speeds Flat bed Automatic send/receive Document loader Roll-feed Ordinary bind copies	$8075 $235/month
Xerox 410 Telecopier	64/96 96/96	4 6	Two speeds Automatic send/receive 75-document loader Roll-feed Electroresistive copy	$2845 $93/month
Graphic Sciences 4000 Series	62 88 124 176	2/4 3/6 4/9 6/12	Multiple speeds Flat bed Automatic 100 document loader Electrosensitive copy Use with FAX 900 broadcaster to transmit to 25 locations	$6240 $159/month + $35-$95/month for Broadcaster
Stewart-Warner Datafax	96	3 (Model 360) 4.5 (Model 240) 6 (Model 180) 9 (Model 150)	One speed Automatic send/receive 75 Document loader Roll feed System Assurance Electrolytic copy Flat bed	$60/month

Two to Six Minute Automated Systems
(Con't)

Device	Resolution (V/H)	Transmission Time (Minutes)	Features	Approx. Cost
Muirhead K4426	63 94 125	2 3 6	Multiple speeds Flat bed Automatic send/receive 200 Document loader Roll feed Electrostatic copy	
Telautograph Copyphones III	83 100	1 3	Two speeds Flat bed Automatic receiver/feed 75 Document loader Roll feed System Assurance Electrolytic copy	
Alden 1100	65	100 seconds	Automatic send/receive Electrosensitive copy Compatible with TV systems	$4000
Alden 1800	96 48	1 3	Automatic send/receiver Electrosensitive copy Copies 15" x 5" or 15" x 18" Compatible with TV systems	$5000
Infolink Scanatron	100/100	1/2 3/6	Two speed modes Flat bed Automatic send/receive 50 Document loader Roll feed Electrolytic copy	$1600

Sub-Minute Automated Systems

3M Company Express 9600	96/200	20-35 seconds at 9600 b/s	Flat bed Totally automated Electrostatic copy Data compression by Huffman Coding	
Graphic Sciences DEX 5100	96/200	20-35 seconds at 9600 b/s	Flat bed Totally automated Data compression by Huffman Coding	$12,000 $300/month
Rapidfax 100	67/200 100/200 200/200	35 50 90	Flat bed Automatic send/receive 50 Document loader Roll feed Electrostatic copy Broadcast to 14 receivers	$265-$285/ month

As can be seen in Table 6, most of the machines are quite similar within each category, varying somewhat on the number of selectable speeds and resolutions. Almost all of the units are transceivers, except for the Graphic Sciences 4000 series of transmitters, receivers, and transceivers, and Infolink's Scanatron model, which is a separate transmitter and receiver. There are, however, some units that have unique features. These include the Graphic Sciences Broadcasters, Xerox Telecopier 200, Xerox 400-1 Telecopier and 3M 2346 machines, and the Alden models.

The Broadcasters made by Graphic Sciences are especially designed for multi-point networks. The manual DEX Broadcaster can simultaneously transmit information to ten locations. Automatic machines in the 4000 series become multi-point systems when interfaced to a FAX-900 Broadcaster, which is a push-button unit. The 900 Broadcaster and DEX 4100 are capable of sending copies to 9, 17, or 25 other units simultaneously, depending on the configuration selected.

The Xerox Telecopier 200, shown in Figure 13, is the only machine reviewed that reproduces the documents on ordinary bond paper. A modulated laser provides the light source for xerographic printing. The Telecopier, as an example of a totally automated machine, has self-checking electronic logic to control send and receive operations. The unit automatically answers a phone call, sets the proper transmission speed, and sends or receives information. An automatic dialer has a digital clock that can be set to transmit material at any time. The dialer reads a coded phone number, dials it, and sends the copies to other unattended machines. Messages

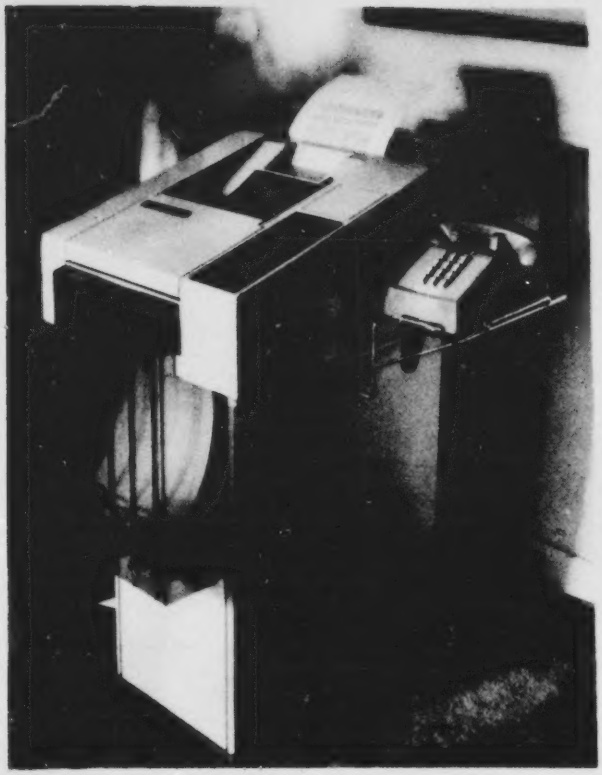

Figure 13. Xerox Telecopier 200 Transceiver

can thus be sent and received at any time without operator assistance.

The Xerox Telecopier 400-1 and 3M 2346 are semi-automatic models. The documents are still loaded separately by hand, but the machine automatically "grabs" and ejects each page. In other manual equipment reviewed, the copy must be loaded onto the drum manually by placing one edge of the paper under a metal clamp-bar. The 3M 2346 machine is shown in Figure 14. Like the Xerox 400-1, it is a desktop transceiver. Four different speeds can be selected; other features include automatic margin adjustment, transmission interrupt, and one-line and error signals.

Alden's automatic machines can also be used to make copies of video pictures, computer printouts, and CRT graphics displays.

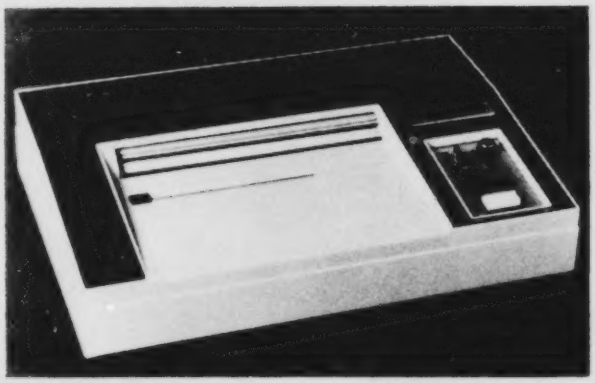

Figure 14. 3M 2346 Desktop Transceiver

Future Developments

Given the limitations of a narrow bandwidth, will it be possible to increase the performance of visual systems much beyond their present capacities? The answer lies primarily in advancements in electronic technology and the potential application of efficient data compression techniques.

As was seen, the main problems encountered by visual systems are the interrelated factors of picture quality, transmission time, and cost. It is relatively easy to improve the quality of the picture by expanding memory storage capacity. A primary concern, however, is the cost of additional memory chips. It is much more difficult to decrease the transmission time, for this involves complex signal processing operations to reduce the total amount of information without overly affecting image quality.

In order to generally assess the future developments of visual teleconferencing systems, this section takes a quick look at some electronic's products that are applicable to videographic systems. It also suggests some data compression techniques that have potential for graphic and pictorial information.

Some Advancements in Electronic Technology

Electronic technology is rapidly and pervasively changing society. Once considered the tool of only scientists and mathematicians, computers are now a part of everyday life. They regulate a car's engine, provide push-button banking, schedule airplane flights, tally our grocery bills, and entertain us with video games. It is not, therefore, surprising that electronics technology is chiefly responsible for the improvements in visual teleconferencing systems.

For the past 30 years, electronic semiconductor devices have continuously been scaled down to make them faster, denser, and to decrease their power dissipation. Since the late 1960's, the electronics industry has spent tens of millions of dollars on research to develop techniques to further improve their components. The results are dramatic breakthroughs in four areas: dense memory chips, charge-coupled devices, one-chip processors, and fast multiplier chips.

At the heart of most of these devices are large-scale-integrated (LSI) circuits and metal-oxide semiconductors (MOS) which circulate the voltage signals carrying the coded information. It is improvements in manufacturing techniques that is increasing their performance while lowering costs. Not only are established methods more efficient, but five new processes have recently been introduced: H-MOS, V-MOS, D-MOS, SOS, and I^2L. (For descriptions of these processes, see Altman, 1977a, 1977b; Le Boss, 1977; Pashley and others, 1977). These techniques are providing electronic capabilities previously not possible, including powerful one-chip processors and memory chips that hold at least sixteen times more bits of information.

Processors: All integrated circuitry is steadily being miniaturized. Computers are a dramatic example. IBM recently introduced a minicomputer which Time labeled "The Miracle Chip." It is smaller than a fingernail, yet its performance is nearly equivalent to a room-size computer of 25 years ago ("The Age of Miracle Chips," 1978).

IBM is not alone in the marketplace of one-chip processors. Products are probably as varied as their applications, ranging from 4-bit microcontrollers that

can be assigned specific functions to 16-bit microprocessors that match the power of general-purpose minicomputers. There are also lines of microcomputers with different performance capabilities. The cost of these chips ranges from $10 for a simple controller to $1000 for a high-performance microprocessor ("Microprocessors," 1976). Fairchild, for example, recently introduced a 16-bit microprocessor that is reportedly one of the most powerful single-chip processors on the market today. Its price of $750 includes a software package and a memory capacity of 32,786 16-bit words. It can serve as many as 63 peripheral devices ("Fairchild 9440 Starts Going," 1978).

The compactness and performance of the one-chip processors make them applicable to almost any piece of equipment, transforming it into an "intelligent" device that is capable of performing many types of functions. In addition, their cost is well within range of even the spare-time hobbyist. These chips are already being used in some visual teleconferencing systems; they can also be applied to others, such as the slow scan televideo. Processors could, for example, perform many signal processing operations.

Memory Chips: Dense memory chips are another area of digital electronics that is rapidly advancing. Today's memory devices are capable of much greater bit-packing density than products introduced as recently as a year ago. In 1977, for example, memory chips that held 4,000 or 16,000 bits became established on the market. This year chips with a 65,000-bit capacity were unveiled, and 262,000-bit products are on the horizon. (Altman, 1977; Koppel & Maltz, 1976; Coe & Oldham, 1977; Coker, 1977). Like processors, there are a variety of memory devices to choose from--dynamic or static random access, read only, programmable or erasable read only, charge-coupled, bubble, and magnetic disc memories. The costs of established chips are steadily declining, and the new high-density devices have potentially an even lower cost.

The availability of increasingly dense and faster memory chips that cost less directly affects those systems that use digital memories to store picture information, including the televideo systems and many of the telegraphics models. It appears that the greater bit-packing density and anticipated lower costs would make it feasible to upgrade storage capacity. The televideo systems, for example, use dynamic random access memories (RAM's) that hold 4096 (4K) bits per chip. The price of these chips is now about 15 millicents per bit. However, the potential cost per bit of the 16k chips, which have four times the storage capacity, should be one-half to one-fourth less, or roughly 8 to 4 millicents (Coker, 1977; Coe & Oldham, 1977; Altman, 1977c). In addition, most of the 16k chips are compatible with the 4k's, so can be used in one system. The new 65k chips are even more remarkable. They make it possible to put all of the memory of the Robot slow scan system on one board, where it now has sixteen.

Charge-Coupled Devices: Charge-coupled device (CCD) memories are another possible approach to increasing storage capacity. They offer a higher bit density and potentially lower cost than RAM's, and they are uniquely applicable to analog systems.

CCD memories are capable of storing 10 to 100 times more bits per chip than semiconductor devices, an important consideration in a slow scan or telegraphics system, where increased picture quality requires multiple levels of storage. Memory chips with 9k, 16k, and 65 k capacities have already been marketed, and future designs will contain 128k bits and more. In addition, the potential cost of CCD memory is 3 millicents or less per bit, about one-fifth the cost of a 4k RAM (Altman, 1977, 1977c).

Because a CCD memory is basically analog in nature, it seems to be particularly applicable to audio and video systems. Amelio (1974) suggests that a CCD could be ideal as a time-sampled shift register. It acts as a variable delay line (i.e., the charge packets are shifted through all the elements of the structure over a desired interval of time) that can be clocked to perform buffer functions. For example, a CCD shift register can be loaded whenever the data appear in bursts (as when a video image is grabbed into memory). The data can then be stored for a given interval and read out at the desired rate. This has obvious applications in a scan converter, for the data can be buffered in and out of memory at fast and slow scan rates. It is also possible to combine RAM's and CCD's in a memory design especially suited to a slow scan system (Metgzer, 1975; Altman, 1977d).

The basic analog nature of charge-coupling has wide application in complex signal processing operations needed for data compression, conversion, filtering, modulation, and multiplexing (Amelio, 1974; Altman, 1974; Gundlach, 1977; Mattara, 1977; Zimmerman & Barbe, 1977). It is expected that a variety of CCD products will be marketed in 1978, including analog-digital converters, delay lines, fast multiplier chips for algorithmic transforms, band pass and low pass filters, and modems. Reticon Corporation has already introduced a CCD transversal filter capable of performing the chirp Z algorithm to calculate a 512-point Fourier transform that can be used for data compression.

A charge-coupled device can play another role--it acts as a natural image sensor that determines the precise location of each element in a picture (present scanning equipment is not exact because it is affected by random electrical noise). As Gundlach (1977) and Altman (1977d) state, CCD sensors can be used in facsimile systems, video cameras, and other imaging products.

CCD technology, which was pioneered by Bell Laboratories in 1970, has been extensively researched by the electronics industry (including Fairchild, RCA, IBM, Bell Northern, Texas Instruments, Intel, Signetics, Honeywell and Phillips. It is based on the relatively simple principle that all mobile electric charge stored in a semiconductor storage element (well) can be transformed to an adjacent well by the external manipulation of voltage. The amount of the electrical charge can represent information. Because the amount of stored charge depends on the applied voltages and capacitance of the storage element, the information can be easily manipulated. In signal processing and memory devices, an input signal represented by a charge packet, is introduced by a standard element and then transferred from well to well by pulsing the appropriate electrodes in sequence.

Fast Multiplier Chips: Fast mulitplier chips are paving the way for complex signal operations that previously could only be carried out by using large off-line computers. These new chips can quickly compute autocorrelations and transforms necessary for digital filtering and compression.

Among the many high-speed chips being introduced, two are described by Waser and Peterson (1977) and Mick and Springer (1976). They perform fast multiplications that make it possible to implement complex mathematical calculations. Both devices are considered to have great potential for digital video compression, voice

compression, and facsimile.

Data Compression

Data compression is apparently essential if good quality images are to be transmitted quickly over a voice-grade telephone line. As electronic technology advances it is becoming possible to implement sophisticated compression techniques that reduce the total number of bits transmitted. This can be done by hard-wired logic, processors, or both.

Hard-wired logic refers to integrated circuitry that is able to perform certain operations without the assitance of an external processor. For data compression, the circuitry must be capable of fast multiplications. In the past, multiplication was performed by a series of additions that required large bit words, were slow, and entailed expensive circuitry. The new fast multiplier chips can now perform these functions.

Microprocessors or micro- and minicomputers can also be used for compression operations. Within a matter of nanoseconds, they can calculate sines, cosines and other complex calculations.

The primary objective of data compression is to transmit only enough bits of information to recognizably represent the original message. The total amount of bits is the product of the number of samples taken and the number of bits used to code each sample. For instance, to completely digitize a message with an audio frequency of 3 kHz, a theoretical minimum of 6000 samples per second are needed (the Nyquist rate). In addition, 7 bits may be used to code each sample, resulting in a total of 42,000 bits of information. It would take roughly 20 minutes to transmit this information over an ordinary telephone line. To reduce the total bit content, the number of samples can be decreased and/or a more efficient coding method can be used. Because a message--graphics, picture, or voice--contains redundant information, it can be adequately represented by less bits.

There are many types of data compression techniques; some are still in the theoretical stage, but many are being applied in both experimental and actual situations. Some of these techniques are:

Two-tone images:	Run length coding
	White block shipping
	Variable length coding
	Predictive differential quantizing
Half-tone images:	Differential pulse code modulation (PCM)
	Differential PCM with adoptive prediction
	Adaptive differential PCM
	Fast discrete cosine transfer
	Delta modulation
	Adaptive delta modulation
	Two-component code
	Interframe coding
	Interpolative picture coding
	Shift Vector transmission
	Linear predictive coding
	Important points

Descriptions of these methods can be found in "Image Bandwidth Compression" (1977) and in numerous other sources.

All of the techniques attempt to eliminate redundant image content by minimizing the number of bits needed to code the information. In most cases, the coding scheme is designed so that only the changes in picture elements are transmitted. To illustrate this process, simple hypothetical applications of two techniques are briefly described: run length coding to compress two-tone images and differential PCM to compress pictures.

Run Length Coding: Two-tone images, such as graphics and facsimile, can be more easily compressed than pictures, for they contain areas that are either all black or all white. In most cases, only two bits are required to code each sample: black is 1, white is 0. Images that contain complex graphics and typewritten text still require a great number of bits to reproduce the total image content and retain satisfactory resolution. Run length coding reduces the total bits transmitted by coding only the changes from black to white, or the locations of the boundary points. For example, a line of picture elements may contain 10 white elements, followed by 4 black, 15 white, and so forth. Rather than coding each element, only the "run lengths" of 10, 4, and 15 are coded and transmitted.

A simple design that employs this coding method might include an analog-digital converter, an encoder, decoder, and digital-analog converter. After the analog signals representing the original image had been digitized, they would be encoded according to a run length format. All the white elements would be represented by a 0 bit. This would not change until a black element was scanned. The encoder could be a processor or a fast multiplier chip. The compression ratio would depend on the level of redundancy in the image and on the particular statistical code used.

At the receive end, the data would be decoded. If the image had been highly compressed, it is possible to enhance, or artificially reconstruct part of the information by a fast Fourier transform. The transform relates the encoded bit pattern to the frequencies and amplitudes of the original analog signal by performing a series of sinusoidal calculations. A fast multiplier chip or processor could be used for this operation. The data would then be reconverted to analog form for display on a monitor or facsimile reproduction. Because a minimum number of bits would be transmitted to represent the image, it may be necessary to also send an error correcting code. If bits are lost during transmission, the message would then be retransmitted.

Huang (1977) presents a detailed discussion of compression techniques for two-tone images, including run length coding. From his studies, he found that Huffman coding resulted in a 5:1 compression ratio for weather maps. Huffman is one of the most efficient variable-length codes. It uses short code words for frequent messages and long ones for less frequent messages. Huffman coding is used in the new sub-minute facsimile machines, such as the 3M Express 9600.

Differential Pulse Code Modulation: Differential pulse code modulation (DPCM) is one of the most widely used methods of picture compression for both narrowband and wideband channels. Like the run length technique, it transmits only changes but the coding method differs. In standard DPCM, a coder predicts the value of the present sample according to information that was previously sent. The difference between the predicted and actual value is then quantized and transmitted. This technique reduces the total number of bits needed to code each sample.

In comparison to black and white images, pictures contain much more information, for they also have gray or color values, and in some cases, animation. It is therefore much more difficult to compress pictures without distorting the quality.

For greater compression, DPCM is often used in combination with other techniques. One method is a fast Fourier transform, which mathematically divides a complex signal into its base frequency and associated harmonics in phase relationship.

In a simple example of picture compression, the original analog signals are first mathematically transformed by a fast Fourier analysis. This transform yields a number of coefficients that represent the base frequency and harmonics. This operation could be performed by a fast multiplier chip, processor, or a type of Fourier filtering device. The frequency coefficients are then coded by a technique such as DPCM to optimize the use of bits. Those coefficients representing less information may be coded by fewer bits, with a greater proportion of bits devoted to the high-content coefficients. Because the high frequencies carry little picture information, many of the coefficients may not be coded at all, compressing the data by perhaps one-half. At the receive end, the signals go through a reverse process; first being decoded and then inversely transformed to convert the frequency coefficients into video signals that can be shown on a monitor. As in the case of run length coding, the Fourier transform can also approximately reconstruct the information that was not transmitted. The efficiency of this compression scheme depends partly on the number of bits needed to code the frequency coefficients and the type of DPCM used.

Differential PCM often results in a 2:1 compression ratio. NEC's color televideo system, for example, cuts the information by half. By combining DPCM with other techniques, the data may be reduced further (Armstrong, 1974; Erikson, 1975). The recent introduction of new DPCM coding and modifications of the Fourier transform may also improve the efficiency of these techniques (Sharma & Netravali, 1977; Zschunke, 1977; Chen & Smith, 1977).

Summary: Run length coding and differential PCM are only two of many techniques that can be used to compress information and decrease transmission time. There are, for example, methods that divide a picture into small blocks for more efficient coding; adaptive sampling rates that conform to the degree of detail in an image and sample more finely in some areas than in others; and techniques that incorporate the psychovisual properties of human vision. As research continues to progress and more coding methods are tested, it is expected that data compression will become increasingly applicable to visual teleconferencing systems. This possibility seems especially promising when one considers the advancements in electronic technology and their pertinence to signal processing operations and the improvement of picture quality.

The growing application of electronic devices to the telecommunications industry and the telephone itself, the trend toward digitizing audio and video information for both narrowband and wideband channels, and the amount of research on data compression all point to future improvements in telewriting, televideo, and facsimile systems.

References

"Age of the Miracle Chip," Time, Vol III, No. 8, Feb. 20, 1978, pp. 44-59.

Altman, R. "Charge-Coupled Devices Move in on Memories and Analog Signal Processing," Electronics, Aug. 8, 1974, pp. 91-101.

_____. "Five Technologies Squeezing More Performance from LSI Chips," Electronics, Vol. 50, No. 17, Aug. 18, 1977a, pp. 91-93.

_____. "New MOS Processes Set Speed, Density Records," Electronics, Vol. 50, No. 22, Oct. 27, 1977b, pp. 92-94.

_____. "One-Chip Controllers and 4-k Static Rams Star," Electronics, Vol. 50, No. 22, Oct. 27, 1977c, pp. 96-114.

_____. "CCDs Show Influence on Design Quietly," Electronics, Jan. 20, 1977d, pp. 22-23.

Amelio, G. F. "Charge-Coupled Devices," Scientific American, Vo. 230, No. 2, 1974, pp. 23-31.

Armstrong, L. "Problem: Cut Video Bandwidth," Electronics, Vol. 47, No. 15, July 25, 1974, pp. 69-70.

Chen, H. W., and C. H. Smith. "Adaptive Coding of Monochrome and Color Images," IEEE Transactions on Communications, Vol. 25, No. 11, Nov. 1977, pp. 1285-1292.

Coe, H. E., and W. G. Oldham. "Enter the 16,384-bit Ram," Electronics, Feb. 1977, pp. 115-121.

Coenen, L. C., Jr. "Interactive Graphics," Plant and Print, 1976.

Cohen, W. C., and S. L. Meyer. "Development of the Educational Uses of Slow-Scan Televideo," Biosciences Communications, 1, 1975, pp. 169-183.

Coker, D. "16-k RAM Eases Memory Design for Mainframes and Microcomputers," Electronics, April 29, 1977.

Cole, B. "IC Houses Eagerly Eye Phone Market," Electronics, April 1, 1976, p. 65.

"Comsat Shows Digital TV Link," Electronics, Vol. 50, No. 16, Aug. 4, 1977.

Erikson, A. "Europe Set for First Digital-TV Step," Electronics, May 29, 1975, pp. 82-83.

Erikson, A. "Digital TV: When, Not If," Electronics, Vol. 50, No. 13, June 23, 1977, pp. 94-95.

"Facsimile: 1978 Overview," Communications News, Vol. 14, No. 12, Dec. 1977, pp. 67-70.

"Fairchild 9440 Starts Going to Customers," Electronics, Vol. 51, No. 1, 1978, pp. 56-57.

Gundlach, R. "Telephones at 100: Transformed by Technology for New Functions," Electronics, Vol 48, No. 25, Dec. 11, 1975, pp. 90-100.

Gundlach, R. "All Technologies Aim at Greater Channel Capacity," Electronics, Oct. 28, 1976, pp. 108-114.

Gundlach, R. "Large Scale Integration is Ready to Answer the Call of Telecommunications," Electronics, April 28, 1977, pp. 93-108.

Heiserman, D. L. "Facsimile Via Telephone," Electronics World, Dec. 1971, pp. 35-38.

"High-Speed Fay Meets Proposed Spec," Electronics, April 28, 1977, pp. 34-35.

Huang, T. S. "Coding of Two-Tone Images," IEEE Transactions on Communications, Vol. 25, No. 11, Nov. 1977, pp. 1406-1425.

"Image Bandwidth Compression: A Special Issue," IEEE Transactions on Communications, Vol. 25, No. 11, Nov. 1977, pp.

Ingram, D. "An Introduction to Slow-to-Fast Television Converters." Amateur Radio, August 1976, pp. 44-45.

Koppel, R..H., and I. Maltz. "Predicting the Real Costs of Semiconductor-memory Systems," Electronics, Vol. 49, No. 24, Nov. 25, 1976, pp. 117-128.

Krallinger, R. E. "Determining Compatibility and Standards for Graphic Communications Systems," Communications News, Dec. 1974.

Lapidus, G. "Remote Graphics Terminals Decouple the User from the Central Site," Data Communications, March/April 1976, pp. 29-36.

LeBoss, B. "C-MOS Gets a Rise out of LSI," Electronics, Oct. 13, 1977, pp. 65-66.

Louv, R. "The Hesitant Birth of Slow-Scan Video," Video Systems, July 1977, pp. 32-39.

Machover, C. "Graphics Displays: Factors in Systems Design," IEEE Spectrum, Vol. 14, No. 10, Oct. 1977, pp. 22-27.

Martin, J. Telecommunications and the Computer, 2nd Edition, Prentice-Hall, Inc., Englewoods Cliffs, NJ, 1976.

Mattera, L. "CCDs Edge Toward High-Volume Use," Electronics, March 17, 1977, pp. 74-75.

Mick, J. R., and J. Springer. "Single-Chip Multiplier Chip Expands Digital Role in Signal Processing," Electronics, Vol. 49, No. 10, May 13, 1976, pp. 103-108.

"Microprocessors: Designers Gain New Freedom as Options Multiply," Electronics, Vol. 49, No. 8, April 15, 1976, pp. 78-100.

Metzger, J. "CCD's Add Flexibility to Memory System Design," Electronics Products, July 1975, pp. 29-33.

Meyer, S. L. "Controlled-Scan Televideo: Efficient Utilization of Communications Channels for Teleconferencing and Data Access," The Telephone in Education, Book II, University of Wisconsin-Extension, 1976, pp. 58-68.

_____. "Telecommunications and the Expansion of Information Services," Conference on Telecommunication Technologies, Networking, and Libraries, National Bureau of Standards, Gaithersburg. June 3, 1977, Northwestern University, Evanston, Ill.

Meyer, S. L. "Extending the Reach of the University with Narrow-Band Telecommunications," Working Paper, Northwestern University, Evanston, Ill., undated.

_____, and D. Brown. "A Review of Available Technology for Narrow-Band Transmission of Visual Material," Biosciences Communications, 2, 1976, pp. 38-48.

Pashley, R., and others. "H-MOS Scales Traditional Devices to Higher Performance Level," Electronics, Aug. 18, 1977.

"Plasma Display Technique Challenges CRT Terminals," Communications News, 1977, p. 46.

Robertson, C. L. "Simple Graphics Terminal," Amateur Radio, 1973, pp. 116-123.

Sakrison, D. J. "On the Role of the Observor and a Distortion Measure," IEEE Transactions on Communications, Vol. 25, No. 11, Nov. 1977, pp. 1251-1266.

Sharma, D., and A. N. Netravali. "Design of Quantizers for DPCM Coding of Picture Signals," IEEE Transactions on Communications, Vol. 25, No. 11, Nov. 1977, pp. 1267-1274.

Smith, J. "New ITT Unit has Eye on the 1980's," Electronics, Sept. 29, 1977, pp.

Steber, G. R. "SSTV to Fast-Scan Converter," OST, March 1975.

"Telecom Changes Could be Too Fast," Communications News, Nov. 1977.

Turner, L. "Which Data-Terminal Display: Plasma Panel or CRT?" Electronics, Feb. 17, 1977, pp. 91-96.

Waser, S., and A. Peterson. "Real-time Processing Gains Ground with Fast Digital Multiplier," Electronics, Sept. 29, 1977, pp. 93-99.

Zimmerman, T. A., and D. F. Barbe. "A New Role for Charge-Coupled Devices: Digital Signal Processing," Electronics, Vol. 50, No. 7, March 31, 1977, pp. 97-100.

Zschunke, W. "DPCM Picture Coding with Adaptive Prediction," IEEE Transactions on Communications, Vol. 25, No. 11, Nov. 1977, pp. 1295-1301.

SLOW SCAN TV TELECONFERENCING

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The still image is universally used for person to person communications. This is true of the printed page, photography, and even television. Visualize, if you will, a typical meeting or classroom situation. Someone speaks to you and elaborates on a theme through use of slides, view-graphs, flip charts, drawings, diagrams, photos, or other means of presenting non-moving pictures. Very seldom is it ever found necessary to introduce actual moving imagery in the form of film clips or video tape recordings.

In the case of teleconferencing, the transmission of single pictures can provide enormous savings in channel capacity, compared to the use of conventional video which generates 30 highly redundant pictures per second. When only audio circuits are readily available, which is usually the case, slow scan TV technology provides a flexible and easy to use way of getting pictures from point A to point B.

A variety of slow scan TV picture transmission systems have been developed over the last two decades; however, the present trend is to use conventional CCTV cameras, monitors, and other system components in conjunction with "scan conversion" devices which reduce the bandwidth of a CCTV camera output from a nominal five megaHertz to approximately one kiloHertz for transmission over voice grade circuits. This is a compression ratio of 5000 to 1, and is generally achieved by a) stretching out the signal in time from 30 pictures per second to perhaps 1 picture in 100 seconds, and b) sacrificing some resolution in the final image.

The slowness of the picture transmission means that a second scan converter with a video memory must be used at the receiving location in order to rebuild clear, flicker-free signals in standard TV format. Once reconverted, the picture may be treated as conventional CCTV and monitored, switched, recorded, or otherwise handled for the user's convenience.

A simple, basic, one-way slow scan system will consist of the following elements:

Transmitter

1. CCTV camera (for image pickup)
2. TV monitor (for assessment of picture produced by the TV camera)
3. Scan converter (for reduction of the signal from the TV camera to audio frequencies)
4. Modem (for converting the slow scan TV signal to a form suitable to the telephone line, typically an amplitude modulated or frequency modulated audio tone)
5. An interconnection to the dial-up phone system (such as an acoustic coupler or, preferably, a Direct Access Attachment or DAA)

Receiver

6. Interconnection to the dial-up phone system
7. Modem (for demodulation of the transmitted signal)
8. Scan converter with memory (for reconversion of the signal to normal TV rates)
9. TV monitor (for viewing of the reconstructed image)

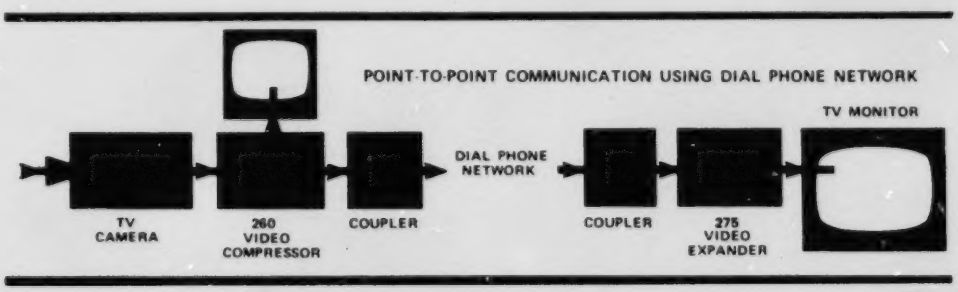


Figure One
Block Diagram of One-Way Slow Scan TV System

Typically, modems are built into the scan converter electronics, thus simplifying system appearance. In essence, a "black box" connects the CCTV camera to the phone line at the transmitting end of the system and a second "black box" connects the phone line to a TV monitor at the receiving end. The main consideration is that the user must now realize that he can no longer rapidly change pictures, as in the case of normal CCTV, but must wait perhaps 60 seconds before a complete picture will be reproduced at the receiving location.

Operation of a simple, one-way system will normally be as follows:

1. The operator places a telephone call to the second location, establishes contact and verifies that communication is desired.
2. Following normal conversation, a picture is transmitted by:
   a. switching the circuit at each end of the system from voice to the scan converter terminals
   b. focusing the CCTV camera on the desired subject
   c. initiating a picture transmission.
3. The viewer at the receiving location will usually see a picture slowly wipe onto the screen from left to right or top to bottom, depending upon the equipment used. At the completion of the picture transmission, switches may be thrown to restore the circuit to normal voice communications, and discussion resumes.

Unfortunately, the slow scan TV signal will normally occupy the entire amount of bandwidth available in a voice channel, thus interrupting normal conversation. When feasible, it is preferable to have a second audio channel available so that voice communications may be maintained in uninterrupted manner while pictures are transmitted when required. This also allows discussion of partial pictures when the format of the graphic information allows.

Two-way operation simply involves the use of both transmitting and receiving equipment at each location. Note, however, that pictures may not be transmitted simultaneously in both directions over the same circuit and, consequently, the voice channel should be used to establish a convention of who will transmit at a specific time.

Depending upon hardware configuration, the picture to be transmitted may or may not be "frozen". In some kinds of low-cost (but high quality) scan converters, it is necessary that a stationary image be provided from the TV camera in order to avoid geometric distortions in the reproduced picture. This is easily accomplished when looking at normal graphic material, but is a handicap when looking at live people. A useful accessory to the transmitting terminal equipment is the "frame grabber" which allows the operator to "freeze" a single image before transmission. Please note, however, that the frame grabber does not really stop motion, and the subject must be stationary during the 1/30th second interval in which the frame is being recorded, otherwise a blurred picture will result.

Figure Two
Slow Scan TV Teleconferencing at IBM

Time is the essential factor in slow scan TV communications, and the amount of time required to transmit a single picture is primarily determined by two factors: bandwidth of the communications link, and resolution of the reproduced image. The dial-up phone network provides a basic limitation to bandwidth, and, although the useful frequency range of a dial-up circuit may be approximately 300 to 2500 Hertz, it is usually necessary to transmit a DC component in the slow scan TV signal. This is usually accomplished by amplitude or frequency modulation of an audio tone, with the result that the effective bandwidth of the transmitted data is only on the order of one kiloHertz, or about 2000 picture elements per second. Under these conditions, a few approximations are as follows:

128 x 128	picture	elements	= 8	seconds
256 x 256	"	"	= 32	"
256 x 512	"	"	= 64	"
512 x 512	"	"	= 128	"
512 x 1024	"	"	= 256	"

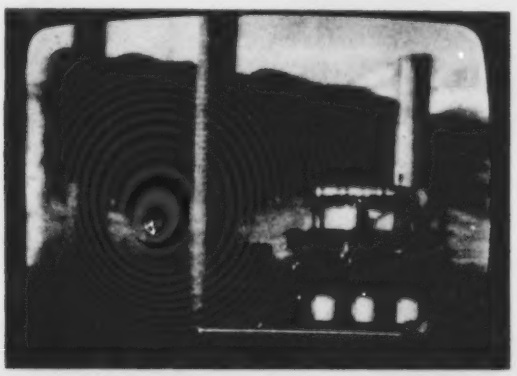
In general, 128 x 128 pictures are too coarse to be useful except in very limited situations. 256 x 256 is acceptable in many instances, while 256 x 512 or higher resolution provides excellent imagery. Key considerations are viewer distance from the TV screen and type of graphic material to be reproduced. In general, the guidelines of conventional TV production may be followed when 256 x 512 resolution is available, while lower resolution will require correspondingly tighter camera shots.

In some instances, wider band communications circuits may be available and scan converters with a faster picture transmission time used. For example, an AM broadcast circuit (100 to 5000 Hertz) would halve the time required for a given resolution, while an FM quality circuit (30 to 15,000 Hertz) can be used to reduce image transmission time by a factor of eight, with room enough left over for a simultaneous voice channel. At present, two mid-Western schools are using the subcarrier channels of their FM stations for distribution of slow scan TV programming, and United Press International will use satellite transmission of 8 kiloHertz slow scan TV pictures to cable companies.

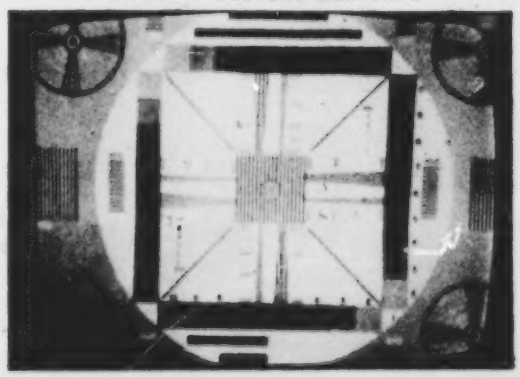
The subjective time required to receive a picture may be minimized by:

- having a separate audio channel so that conversation is uninterrupted

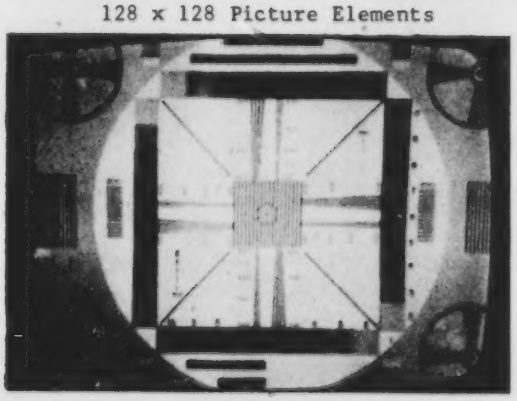
64 x 64 Picture Elements



128 x 128 Picture Elements



256 x 256 Picture Elements



256 x 512 Picture Elements

Figure Three
Slow Scan TV Reproduction

- using equipment configured so that the new picture wipes off the old, and the viewer sees new information almost immediately

- using "split screen" techniques, with only part of the raster being filled with new material

- using a second video memory at the receiving location so that transition from picture to picture seems instantaneous (memories are "ping ponged", with one recording a new picture while the second is providing the viewed display)

- employing multiple memories at the receiving site so that several images may be simultaneously displayed to the audience

- recalling previously transmitted information from tape or disc.

The use of the last two mentioned expedients provides an extra dimension in teleconferencing and can, in some instances, provide communications superior to live TV.

Receiving terminal configurations may also include multiple monitors, large screen projection, and distribution of program material to other locations in the area by cable. The scan converted video and accompanying audio may be video tape recorded for later review or, alternately, recorded on ordinary audio cassettes for playback through the scan converter at a later time.

The transmitting part of the system may also be made more flexible by addition of more cameras. A typical configuration might use one camera with frame freeze for pictures of conference participants, with a second camera focused on flip charts or a blackboard, and a third on a copy stand for viewing of pre-prepared graphics. Slides, film, or video tape may also be used for source material, with a frame grabber usually being desirable for use with the last two. Normal good video practices should be employed regardless of the video source in order to present the best picture to the final user.

Slow scan TV production is simple. Have something significant to say and illustrate it with useful images. Unfortunately, user experience, to date, is largely limited to a relatively small group which includes IBM, Satellite Business Systems, Bell Canada, NASA, the U.S. Office of Telecommunications, and a number of other government agencies. The transmission economies of still frame television make usage likely to expand rapidly in the near future, particularly as more high quality audio channels become available through satellite transmission.

PICTURE TRANSMISSION FOR AUDIO TELECONFERENCING

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Sending and receiving frame-freeze TV pictures over the telephone network is now feasible from the standpoints of cost and transmission time. This capability greatly enhances the potential for information exchange in teleconferencing situations, by adding pictorial information to the normal verbal exchange. While visual data is not the solution to every problem of interchanging information among participants in a conference, it is still true in many situations that "a picture is worth 1,000 words."

System Operation

The operation of a typical system for transmitting and receiving pictures over the telephone is described below. (Some of the compelling reasons why the system operates this way are discussed in the following section).

In a teleconferencing situation, the person wishing to convey pictorial information will place the pictorial material in front of a standard TV camera and cause a scan-converter to which the camera is connected to frame-freeze the subject matter. This can be done without interfering with the verbal exchange of the conference; i.e., setup is "off line." In fact, if necessary, as much time as necessary may be taken to prepare the subject matter, see that it is properly lighted, and inspect the frozen frame in the scan converter's memory (which is simultaneously being displayed on a local TV monitor) to see that it contains the desired information.

When everything is ready, the person transmitting takes control of the teleconference line, announces a forthcoming picture and switches the scan-converter, already connected to the telephone line, to "transmit."

At one or any number of receiving locations, a similar scan converter receives the picture information and places it in its memory. Simultaneously, the information in memory is played out in the form of video for display on a local monitor. In a currently popular system, transmission of each picture takes 8 seconds; the appearance on the receiving end is that of a window-shade being lowered with the new picture on it.

Several options exist regarding single or multiple picture transmission. And control over continuing to receive additional pictures or "holding" a desired picture for extended viewing exists at the receiving end. In addition, the received pictures may be simultaneously recorded on an ordinary audio cassette tape recorder for later playback and inspection (in effect, the tape recorder on playback takes the place of the telephone line).

Principles

Bandwidth Reduction

In order to transmit pictorial or any other information over the telephone network, the first requirement is that the information transfer rate must fall within the limits which the telephone network can tolerate. The maximum information transfer rate of any communications channel is defined in terms of its bandwidth. This in turn is defined by the maximum number of electrical cycles per second that the channel will pass (or more precisely, above which severe distortion of the transmitted information occurs).

A "cycle" can be thought of as one complete excursion of the electrical signal from some starting value, out to another value, and back to the starting point. In terms of picture information, a cycle is therefore roughly equivalent to going from say, black to white and back to black again (or dark grey to light grey and back again). Each point in the picture is defined as a "picture element" or pixel, and we thus see that in any regular scanning system it takes two adjacent pixels to make up one equivalent electrical cycle.

Standard TV requires a bandwidth of about 4 MHz (4,000,000 cycles per second). This can be readily calculated as follows. A TV picture consists of approximately 500 picture elements (pixels) along each horizontal scan line, and approximately 500 scan lines. Thirty of these pictures are transmitted every second. Remembering that it takes two pixels to make one cycle, the product of these numbers gives us 3.75 MHz, a shade under the above figure.

Unfortunately, the standard telephone network has far less capacity than this. With one of the common methods of modulation currently in use, its capacity for distortion-free transmission is about 1 KHz (1,000 cycles per second). Clearly, there is a disparity by a factor of 4,000:1, and something must give.

Since the telephone network is not going to change dramatically in the near future, what must give is the speed and resolution (number of pixels) of the transmitted picture. TV transmits 30 frames per second in order to portray motion. If we give up motion and transmit only still, "frame-freeze" pictures over a relatively long time, a substantial inroad on the problem can be made. If, for example, instead of 30 pictures per second we transmit only one picture over an 8 second period (one popular present standard) then we gain by a factor of 240:1 (1/30 second versus 8 seconds). That's a great improvement, but still not enough. However, if we also reduce the amount of detail in the picture from 500 pixels per

TV line to 128 pixels per line (about 4:1), and from 500 scan lines per picture to 128 lines per picture (another 4:1), we have picked up another factor of 16, and have virtually achieved the 4000:1 compression we need (16 X 240 = 3,840).

This is precisely the basis for at least one popular present system for picture transmission over the telephone network. Clearly, other possible trade-offs can be imagined between resolution (number of picture elements), time of transmission and bandwidth. Some of these will be noted in a later section

Modulation Techniques

Most phone-line TV systems in use today employ Frequency Modulation (FM) for conveying the picture information. It has been found from experience that placing the modulation frequencies in the middle of the voice band of the telephone provides for reliable transmission without interference to other users. A typical standard is one in which 1500 Hz represents black, 2300 Hz represents white, and intermediate frequencies represent intermediate grey shades.

This type of modulation has several advantages. By the nature of FM, the transmitter is supplying a maximum amplitude signal to the channel at all times, thus making the signal as large as possible relative to noise. In addition, variations in the amplitude of the received signal from time to time or from telephone line to telephone line are suppressed over a wide range. This is a particular advantage in recording the picture (on an audio tape recorder) since partial "dropouts" on the tape do not show up in the play-back picture.

The disadvantage of FM is that it is inherently a double-sideband signal, meaning that frequencies both above and below some central frequency ("carrier") must be transmitted for full fidelity of the information. However, it is well known that in Amplitude Modulation (AM) systems, all of the information in the signal is contained in either the upper or lower sidebands, and that only one sideband need be transmitted for full fidelity. This is Single Sideband (SSB). In practice, it is usually advantageous to transmit all of one sideband and only a little (a "vestige") of the other sideband. The result is Vestigial Sideband AM or VSBAM. By taking advantage of this technique, either a narrower bandwidth can be used, or more information per second can be pumped down the same bandwidth, by approximately a factor of two over FM.

The disadvantage of VSBAM is that the signal is no longer maximum amplitude at all times (it is "amplitude modulated") so the signal-to-noise ratio suffers. In addition, since a partial drop-out on a recording tape looks like a change in amplitude that is indistinguishable from a change in amplitude due to picture content (the "real" change), high quality audio recorders and tapes must be used if the play-back picture is not to be marred by these defects.

Trade-Offs

From the foregoing discussion it can be seen that a number of possible trade-offs exist among picture resolution, transmission time, and other features.

Resolution

The limited amount of picture detail that can be conveyed in the 128-line system described above is illustrated in the accompanying figures (Fig. 1 through 4). It leads to the natural question: why not improve resolution? The answer is that of course this can be done, but there are penalties. One is transmission time. To achieve a visually significant "improvement" in resolution translates, based upon common experience, to doubling resolution. But this means doubling both the number of pixels per picture line and the number of lines. Given no change in the characteristics of the telephone line, this means taking 4 times as long to transmit the picture (e.g. from 8 seconds to 32 seconds). In addition, cost of equipment goes up dramatically, since 4 times as much memory is required to store the pictures in the scan converter at each end.

Other Channels

There are, of course, other types of communications channels than the standard telephone line. If these offer increased bandwidth, shorter picture transmission times can be achieved. For example, without exceeding the bandwidth of a high-fidelity audio tape recorder, it is possible to reduce transmission (or in this case recording) time of a 128 by 128 pixel picture to about one second.

Other Modulation Techniques

In the interests of perfect fidelity of picture transmission, it is sometimes desired to transmit the information by digital means. In the example system above, this means that approximately 16,000 picture elements (128 by 128) must be transmitted in 8 seconds. The grey scale value of each picture element is represented by some number of binary bits in the scan-converter memory. While more grey shades are desireable, it has been found that 16 levels are acceptable, or 4 bits per picture element. The resulting digital bit rate is approximately 8,000 bits per second. This is above the capabilities of all but the most expensive digital modems operating over specially-conditioned telephone lines.

Simultaneous Voice

It is sometimes deemed a drawback that the voice portion of the teleconference must be discontinued while picture transmission is in progress. At the present time there is no known means for putting both voice and picture signals on the same channel at the same time. This is because, in the interests of speedy transmission, the picture signal occupies as much of the audio channel as possible. In other words, there is no "room" left for voice, and there is no basis on which to separate the two if they are once mixed. The result would be interference to both voice and picture.

Picture Subject Matter

While the uses to which phone-line TV can be put are virtually unlimited, a word or two of do's

and don'ts with respect to subject matter seem to be in order.

First, do send pictures of things that are difficult if not impossible to describe in words. What things are these? They include simple line drawings, sketches or signatures, to name but a few. Can you imagine trying to convey, in words, an accurate representation of the schematic in Fig. 2, the nature of Mr. Doe's signature in Fig. 3, or the exact shape of the chart recording in Fig. 4?

But do not make the pictures too complex. Because of the limited resolution in most phone-line TV systems, only relatively simple subject matter comes through clearly.

And, generally, do not send pictures of printed or written words. That's what voice is for. You can say, "The forecast for today is early morning fog along the coast and inland clouds," in less than 8 seconds that it would take to send a picture of that printed statement. And with more (or less) conviction.

Figure 1. Personal contact by phone-line TV. An example of a 128-line picture transmitted over the telephone.

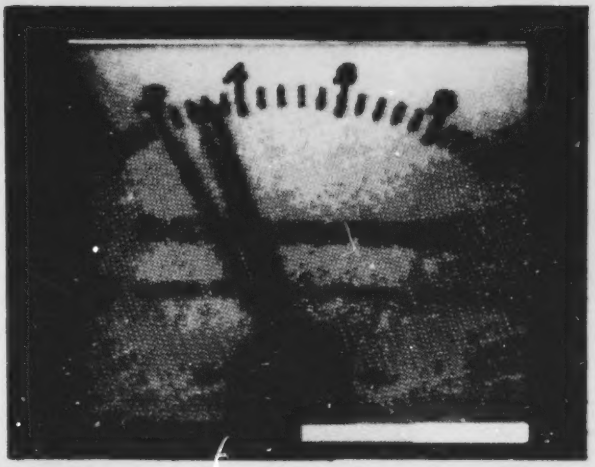

Figure 2. Schematic via phone-line TV.

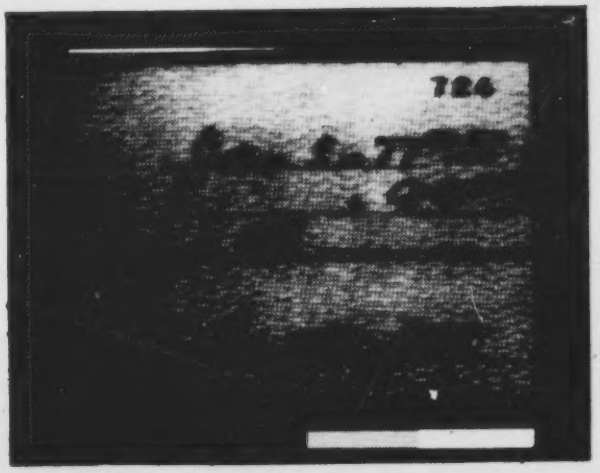

Figure 3. Signature via phone-line TV

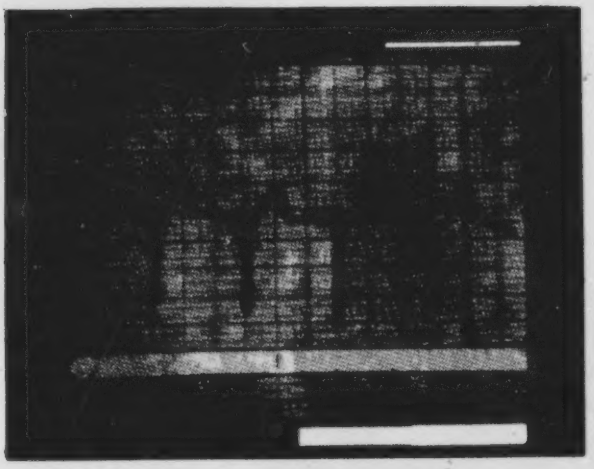

Figure 4. Chart recording via phone-line TV.

TELECONFERENCING AS A COMMUNICATIONS TOOL

Hugh F. Gilloqly
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Teleconferencing is generally defined as an exchange of information and data via electronic methods in a conference setting involving multiple participants at two or more locations. It is intended to approximate, or to serve as an alternate for, direct face to face confrontation. It allows a group of individuals working at separate locations to confer with one another as if they were all seated around the same conference table. As a result, teleconferencing is seen primarily as a substitute for long distance travel to and from business meetings and professional conferences, with travel time and cost savings.[1] Some teleconferencing systems are currently in operation and a few commercial services are available, but in general, implementation is just beginning.

Teleconferencing is a particularly important new use of communications since it can generate significant savings in travel time and cost, energy consumption and response time.

People have been trying to squeeze visual information through telephone lines and other narrow band communication channels for a long time. We, at NEC, now feel that we have the units that are capable of doing this. The units that we refer to are NEC's TVS-754 Telephone Video System in monochrome and our TVS-751 color Telephone Video System. The TVS-754 is designed to be used exclusively with a voice grade telephone line while the TVS-751 is designed to be used with a digital transmission line of 2.4 kilobits up to 1.544 M bits. The TVS-754 is an analog signal transmission system designed for high speed transmission of monochrome pictures. This unit is capable of delivering high quality pictures anywhere there is a voice grade line available in approximately 30 seconds of time. The TVS-754 will operate with standard television monitors and cameras. It does not require any special applications or adjustments in order to be operational. Its use in teleconferencing does require one telephone line for the audio portion and one line for the visual transmission. A connection device into the telephone network is normally required along with a separate telephone instrument with an exclusion key device. These units are available from the local telephone operating company. Conferencing may be set up with as many as 10 or 12 units tied into one main transceiver. However, in a conference of this size, it would require some type of scenario in order to be an effective communication tool. The output of the TVS-754 may be stored on good quality audio cassette for storage purposes and/or played back at a later time. This audio cassette would have to be processed back through a TVS-754 and displayed on a normal television monitor.

Basic Description of TVS

NEC has developed the Telephone Video System or freeze picture transmission system which is capable of transmitting a video picture through a narrow band transmission line. We feel that this system offers wide applications in the fields of teleconferencing, medical diagnostics and documentative transmission. The system employs A to D converters and storage in IC memory, this assures high quality pictures and eliminates the need for adjustments. It also provides simplified maintenance.

As a transmission system either the analog system or the digital system may be used. The analog system is designed for monochrome transmission while the digital system is designed for the transmission of color information. NEC manufactures transceivers which have the capability of transmitting and receiving as well as transmit only or receive only depending upon the practical needs and applications. Options are available for either one of these systems to have dual memory capability for full duplex transmission, pointer generator options and scrambling techniques are available on the digital system.

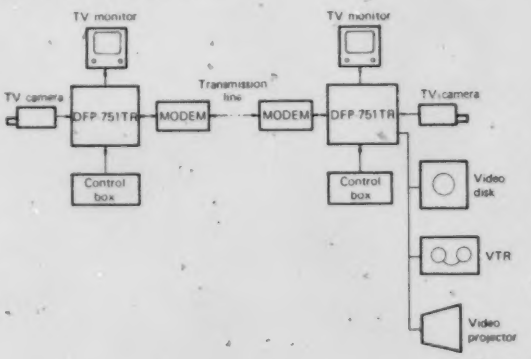


Fig. 1 Block diagram of the freeze-picture transmission system.

The TVS-751 color system utilizes digital transmission for high quality pictures. Figure 1 shows the block diagram of TVS-751 TR which is a transmit and receive unit. The most prominent feature of the TVS-751 is the use of HO-DPCM (Higher-Order-Differential-Pulse Code-Modulation System). This system is capable of converting the composite NTSC color signal directly into DPCM without separating the luminance signal and chrominance signal, reducing the 8 bits/sample to 4 bits/sample. The use of HO-DPCM has made it possible to reduce the transmission time to one half that of the linear PCM system while also reducing the capacity of the picture memory by half.

In the TVS-751 the video input signal is converted into 8 bits/sample linear PCM through the A to D converter. This 8 bits/sample signal is reduced to 4 bits/sample by HO-DPCM and stored in the memory which is read out by the clock signal corresponding to the data transmission speed of the transmission line. The parallel signal thus obtained is converted into a serial signal through the transmission controller and is sent to the transmission line after synchronous

code is applied. The signal stored in the memory is read out at a slow rate for transmission and at the same time, it is read out at the TV rate for monitoring the picture.

The signal is again converted into the original analog TV signal by the HO-DPCM decoder and D to A converter so that a still picture signal is fed to a TV monitor. In reception the serial signal from the line is converted into a parallel signal by the signal transmission controller and is stored in the memory - this signal is read out at the TV rate, as in the case of the transmission to output a still picture signal.

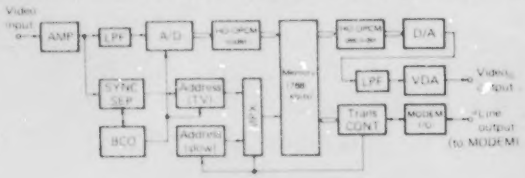


Fig 2 Block diagram of the DFP-751TR

Since the sampling clock for A to D conversion used 10.7 MHZ frequency three times the color sub carrier, the total information of a picture (one picture contains one field of 262 TV scanning lines) is approximately 750,000 bits. Accordingly, the time required to transmit one picture is obtained by dividing the total information by the data transmission speed of the transmission line. Table 1 shows the time required for typical data transmission speed. The TVS-751 does require a modem interface into the telephone facilities and depending upon the type of line and modem governs the transmission speed. The differences between the TVS-751 and 754 is that the 754 uses 7.5 MHZ sampling frequency instead of 10.7 MHZ and it records the signal of linear 8 bits PCM on the memory without using HO-DPCM. That is because of the analog transmission system. The signal is read out from the memory at the slow rate and is converted into an analog signal through the slow rate D to A converter and frequency modulated for transmission. In reception, the demodulated slow video signal is converted into linear 8 bits PCM signal through the slow rate A to D converter and is stored in the memory.

Table 1 Freeze-picture transmission speeds

Data transmission speed	Time taken for transmission of one picture (sec)
1.544 Mbit/sec	0.5
48 kbit/sec	15
9.6 kbit/sec	75
4.8 kbit/sec	150
2.4 kbit/sec	300
1.2 kbit/sec	600

Systems Configuration

In a normal monochrome teleconferencing application for optimum performance we would recommend a minimum of three black and white cameras with manual zoom lenses and one remote control of zoom, 2 monitors, minimum size 17" and for audio, desk type speaker phone that is furnished by the local telephone company. Each location should have a mirror image of the above equipment. Other optional equipment such as audio recorders, video disk recorders and other accessories may be included for the users at their discretion.

The TVS-751 color system would not of necessity require three color cameras, because of the expense involved, we feel that two inexpensive color cameras could be used as well as two color monitors and the appropriate audio connection and other optional accessories as needed.

Teleconferencing Benefits

The communications improvements in organizations is as follows: faster response to problem solving, pre-meeting communication and trip follow-up reports. The organizational structure alternates would be effective people access, communications flow and skills enhancement of personnel. The economic benefits would be travel costs savings, productivity gains and less time loss due to travel.

Optional Characteristics

Some of the findings we have found to be effective is in keeping the equipment in a low profile configuration and not leaning toward typical TV studio presentations. The equipment should be easy to learn and operate, and one person in primary control through a console arrangement. A normal meeting schedule protocol should also apply in teleconferencing. It is also helpful to have introductions of personnel via the video in establishing the meetings. We have also found that minimum discipline is necessary but there should be controls on arrivals and departures from the meetings as far as personnel is concerned.

Media Flexability

The TVS system lends itself very readily to all types of normal media presentations in use today such as flip charts, foils, slides, people images, blackboards, photos, prints and films. All of these above have been successfully integrated into teleconferencing use.

Successful Applications

There has been a wide range of work oriented business conferences found to be successful in teleconferencing such as problem solving, information exchange, project/program review and remote diagnostics of patients in medical applications. The use of product planning as well as failure analysis schedule review and engineering changes are also useful applications in teleconferencing.

Summary

The results of both laboratory based studies and visual studies have indicated that teleconferencing was perceived as most satisfactory for activities characterized by the presentation and exchange of ideas and information. Furthermore, compared to audio only telecommunications systems, video telecommunications was preferred for nearly all activities.[2]

[1]Samuel H. Hebbard, Acting Director Communications Programs, National Aeronautics and Space Administration, SATELLITE COMMUNICATIONS, October 1977.

[2]A. Michael Noll, AT&T, Morristown, New Jersey, TELECONFERENCING COMMUNICATIONS ACTIVITIES

THE TALOS TELENOTE SYSTEM

Talos Systems Inc.
Scottsdale, Arizona

Introduction

This paper describes Telenote[tm] systems in use with the dial-up telephone system. The systems described are categorized by two methods of telephone interface:

1. Coupler Interface Method
2. Direct Telephone Line Interface Method

Telenote[tm] can tie together various departments and remote locations via a simultaneous, multi-point, hard-copy network. Oral communication can also be used to clarify and discuss the graphic material which is transmitted. Selections can enable the user to direct both written and oral communications to just one or any number of receiving stations to insure an efficient communication system.

Telenote[tm] is a unique communications system that allows its users to sketch ideas, jot down figures, write notes or fill out a pre-printed form, alternately as he or she talks. The identical graphic or oral information will be received at any one or more receiving point - instantly and simultaneously - exactly as it has been sent out. It is a reliable, legible, and easy to use graphic communication system.

The coupler interface method. (i.e. CIM) is accomplished by simply dialing the party desired on any telephone instrument and inserting the telephone handset into the unit's acoustic coupler. Both voice and graphics are available through the Telenote.[tm] This method most commonly used for non-dedicated line applications.

The direct telephone line interface method. (i.e. DTLIM) is accomplished by physically connecting the Telenote[tm] unit to the telephone system through one of the unit's direct access accessories. This method provides more convenience when the unit is used constantly at a fixed location employing dedicated telephone lines.

General Information

Telenote[tm] is designed to operate over communication lines according to the following general format:

Voice	300-3000	HZ
Data		
X Carrier	1400-2100	HZ
X subcarrier	70-110	HZ
Modulation	FM	
Deviation	±150	HZ

Interfacing to telephones with the Telephone Handset Coupler utilizes acoustic coupling for sending and magnetic coupling for receiving.

Telephone handsets must be capable of magnetic induction pickup of their earphone. This generally requires the use of the following telephone:

Western Electric Type 500 or equivalent

Other telephone types, such as the ITT, are often completely compatible. However, such others as the Stromberg-Carlson are physically compatible, but not capable of magnetic coupling with the earphone supplied in the handset. Telephones supplied by General Telephone are not physically or electrically compatible.

In certain areas, the telephone company will exchange their telephone or the earphone element for one that is compatible, other areas will require utilization of a special accessory that allows the Telenote[tm] to be electrically coupled to the telephone lines in operation with an interconnection device.

The Telenote[tm] family consists of transmitters, transceivers and receivers plus many other accessories. (see attachment)

Some Typical Applications:

A. Diet Change Order System
B. Aircraft Parts Supply System
C. Flight Operations System

Other proven applications are in the area of:

Health services-Hospitals, Clinics, Pharmacies
Government-Military, FAA, Police, Fire Departments
Manufacturing-Production, Sales, Warehousing, Shipping
Processing Plants
Transportation-Airlines, Trucking, Railroads
Hotels/Motels-Housekeeping
Education-Homebound Instruction, ETN (Educational Telecommunications Network)
Sports
Utilities-Dig In (Blue Stake)
Finance-Banking, Insurance, Investment Brokers
Merchandising-Retail, Wholesale

Special models are also available such as a specially equipped Telenote receiver which provides overhead projection, specifically designed for ETN teaching and business teleconferences. A greatly enlarged image

of the graphic material being received on the Telenote[tm] writing surface is projected on a wall or screen for group viewing. The hard-copy graphics received can be reproduced on any office copier. The Telescreen unit also has audio capability, this permitting voice/graphic lectures or conferences over any telephone.

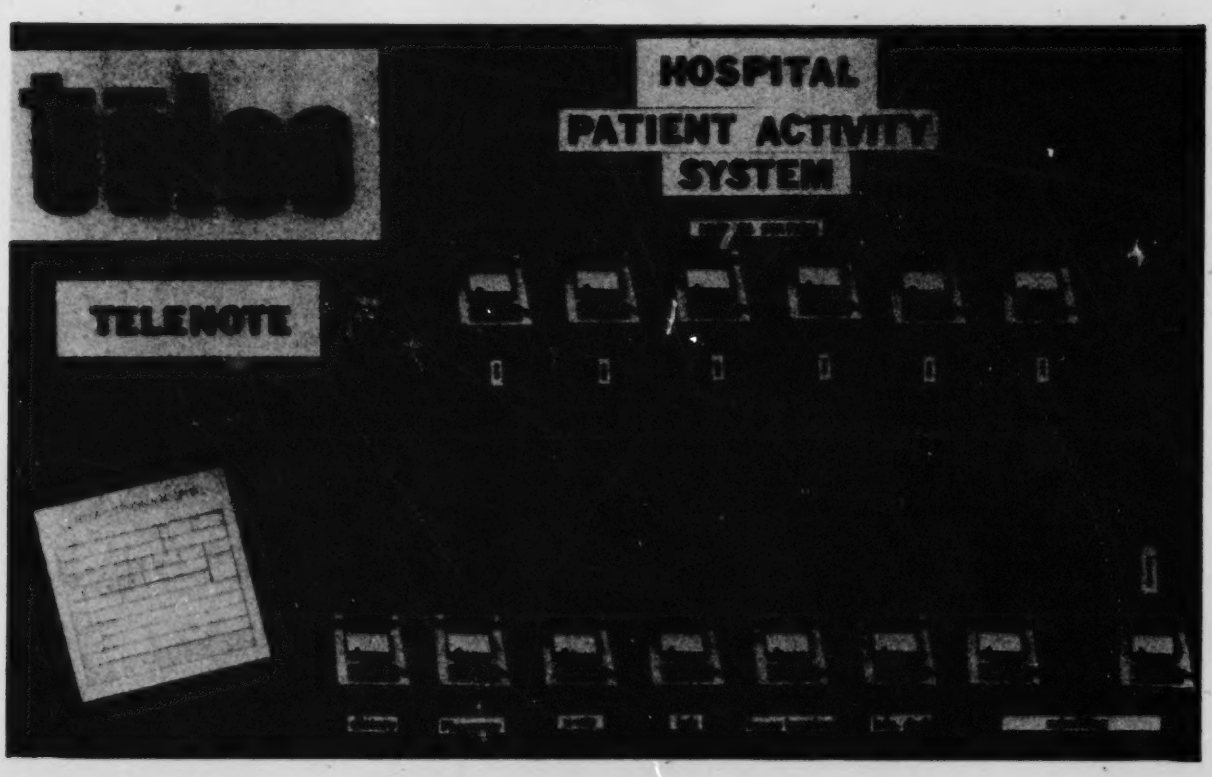
All Talos Systems Accessories are classified as A.C. Communication Components and require communication lines having the following characteristics.

Short Distance-up to 1000 feet. Any pair of 18-20 gauge twisted conductors having insulation material suitable for the specific environment.

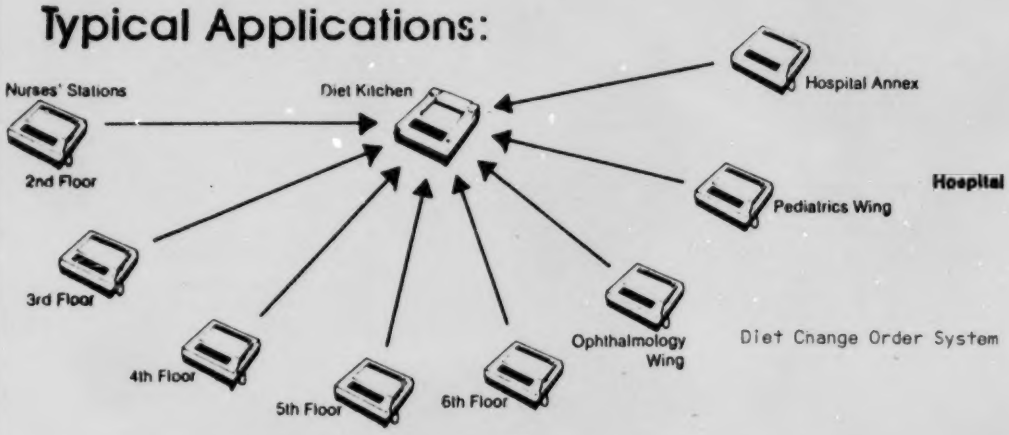
Long Distance-over 1000 feet. Telephone companies private leased lines such as the 2001 Audio Grade Service. The Telenote[tm] has a proprietary communication over ordinary telephone lines without conditioning. It is characterized by high noise immunity, and therefore, can operate in several communication environments.

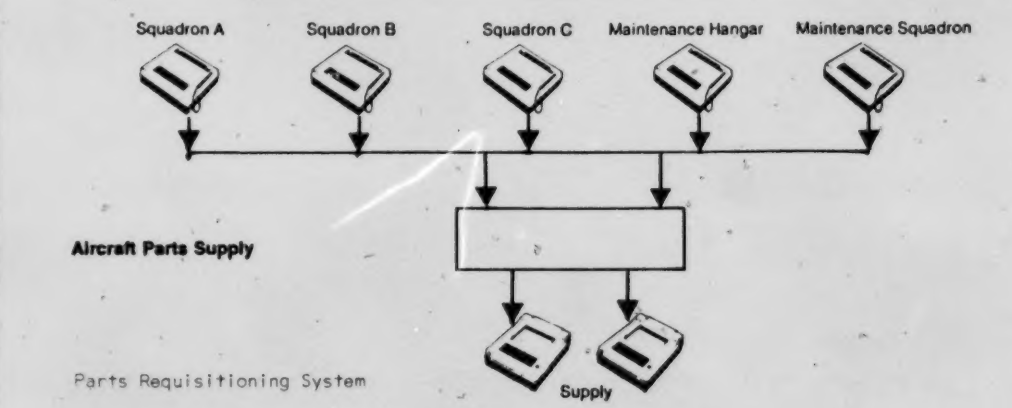
In summary, the Telenote[tm] product line is extremely versatile and flexible. The system performs the functions of communication control, management and interface in operation with communication lines having minimum audio grade capabilities. Options include:

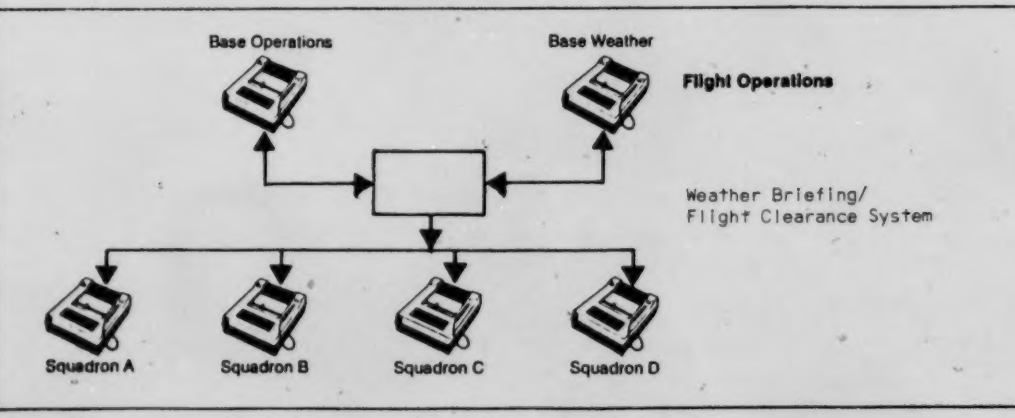
Transceivers	- Transmits and receives, also with telephone couplers and/or conference audio options
Telescreens	- With overhead projector, conference audio and telephone coupler
Receivers	- Receive only with other options mentioned above
Transmitters	- Transmit only with other options mentioned above



Typical Applications:







118

ELECTRONIC BLACKBOARD
HAVE CHALK WILL TRAVEL

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Bell Telephone Laboratories, Incorporated
Holmdel, New Jersey

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Introduction

This paper describes a new communication tool called the Electronic Blackboard. This system is capable of transmitting text and drawings, handwritten with ordinary chalk on a special pressure-sensitive blackboard, over ordinary telephone lines to any other location where the writing can be received and displayed on a television monitor in real time, i.e., simultaneously with its generation at the originating location. The person at the receiving location can write or draw on his (or her) blackboard and send this writing back to the first person's display. Thus, writing can be transmitted in both directions, or if several systems are tied together, it may be transmitted in many directions, providing interactive, multipoint, real-time visual communication. The Electronic Blackboard, therefore, can be a valuable communication tool in education and in teleconferencing.

Telewriting equipment has been available for many years and has been used for various tasks, such as sending messages from a hotel lobby, writing sales orders, and confirming car rental reservations. Some of the systems have also been used for remote teaching and conferencing, with various degrees of success.

Although it is not the purpose of this paper to discuss the features and limitations of existing telewriting systems, it is desirable to enumerate, briefly,

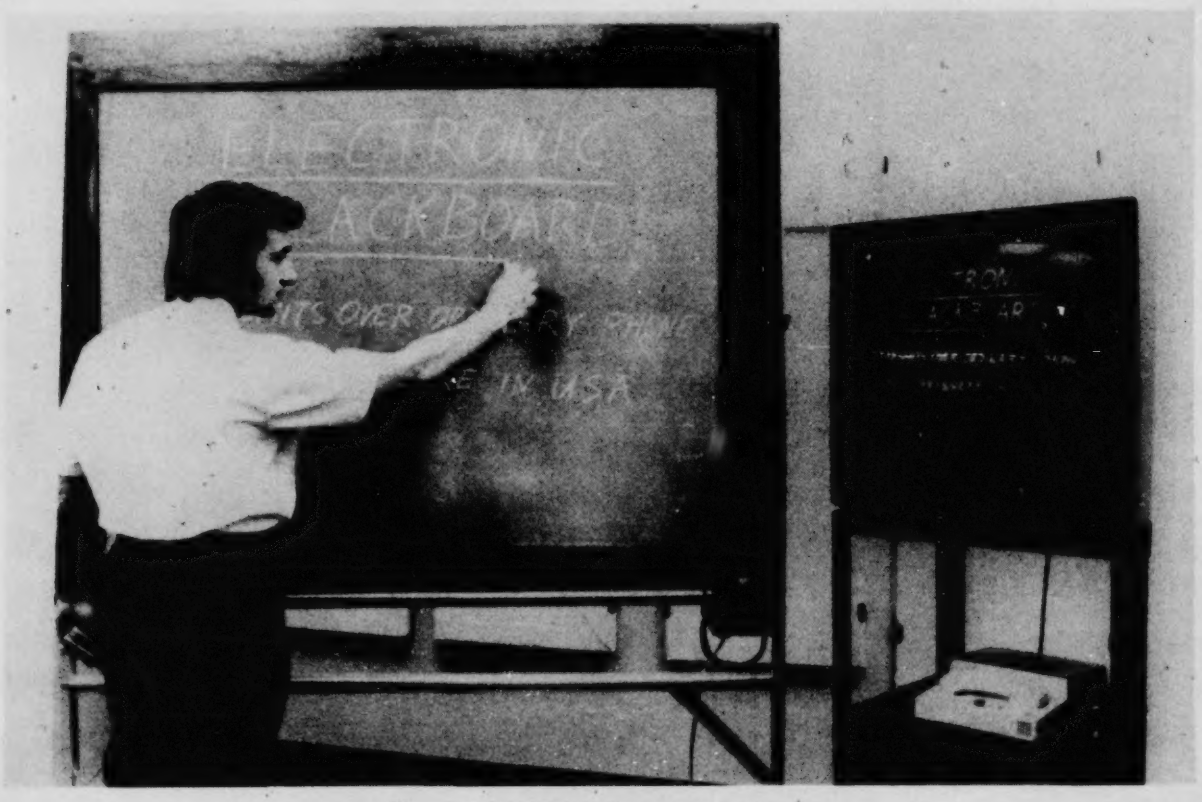

Fig. 1 - Gabe Torok demonstrates the electronic blackboard and monitor.

the human factors requirements of an ideal telewriting system.

An ideal telewriting system designed for remote teaching and teleconferencing should require very little training to provide effective service. It is essential that the user become comfortable with the system and be able to forget, within minutes, that he is using a sophisticated electronic apparatus. Otherwise, user acceptance will be limited to a few "technology-oriented" mavericks, and wide acceptance of the new service will be hampered. This dictates that the environment in which the teaching or conferencing takes place be familiar and uncomplicated. The use of the input terminal should be simple, and free of mechanical or electronic contraptions, at least as far as the user can see. Control switches, lamps, and buzzers should be kept to an absolute minimum even at the expense of limiting some features to the user. The display should provide a good quality, high contrast image and should be of sufficient brightness to minimize the need for special treatment of room lighting. The system resolution should be such that restrictions on what the user wants to write or draw at the sending end are minimized.

In addition to the requirements for the visual part of the communication it is essential that good quality audio accompany the graphics and that the system preferably provide hand-free telephony in sychronism with the accompanying graphics.

If the system is used on the switched telephone network, establishing the graphics connection should not entail anything more complicated than placing a telephone call. Finally, the system must be virtually maintenance free. Nothing is more discouraging than to start a viewgraph presentation only to discover that the projector bulb is burned out and a spare is not readily available.

What follows is a detailed description of the Electronic Blackboard system. In addition to the description of how the system realizes the above requirements, it will be pointed out where cost and engineering factors resulted in compromises which make the Electronic Blackboard, nevertheless, a highly effective communication tool.

Figure 1 shows a complete send and receive terminal of the system. The two-way voice communication takes place on a separate telephone line using a 50A1 Portable Conference Telephone Set.* This device has a loudspeaker, a built-in microphone, and portable lavalier type microphones providing hands-free operation.

Input Terminal

The input terminal of the Electronic Blackboard (Figure 2) is a pressure-sensitive blackboard, 1.3 meters by 1.6 meters in size. It consists of two large and very flat sheets separated by a 1.5 mm air gap. One of the surfaces is rigid, the other is flexible. The flexible sheet is coated on the outside with blackboard paint on which the user writes with ordinary blackboard chalk. The facing surfaces of the two sheets are electrically conductive. When writing occurs the flexible membrane deflects slightly under the pressure of the chalk and makes contact with the rigid sheet. The tension in the flexible membrane is fairly uniform and a 50-gm force on the chalk is sufficient to make the contact. Since insufficient pressure would result in skips on the display, writing must be somewhat more deliberate than on an ordinary blackboard.

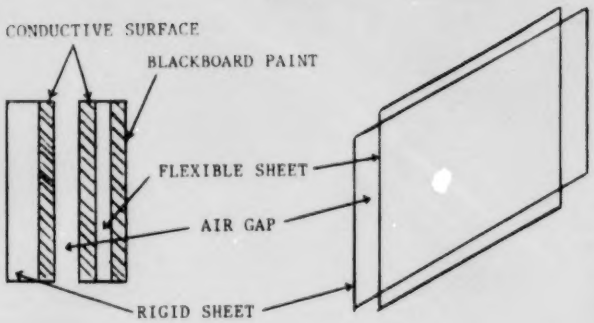

Fig. 2 - Blackboard

The position of the chalk on the blackboard is detected by a simple and inexpensive analog voltage division technique. Time division multiplexing is used to detect the voltages corresponding to the X and Y coordinates of the chalk's position. The resolution of the blackboard is essentially infinite because of the continuous nature of the conducting surfaces.

Practical considerations, such as flatness, tension, weight, and transportation dictated that the blackboard's size be limited to 1.3 meters by 1.6 meters. In order to provide more board space, such as might be desirable for a graduate level engineering course, provision has been made to connect one, two, or three blackboards to the system. The modular concept permits custom tailoring an installation for a specific need. Partial erasing of the board can be done with the corner or tip of an ordinary blackboard eraser. Lifting of the eraser from a special eraser tray causes an infrared light beam switch to be activated. When the eraser touches the board an erase command rather than a write command is generated. The user must replace the eraser on the tray before writing is resumed. Since the system can only detect one X-Y position at a time, only one small spot at a time can be electronically erased unambiguously. When the entire board is to be cleaned the board is erased conventionally with the eraser and a CLEAR pushbutton on the eraser tray is pushed to send a CLEAR command to the display. This is the only switch that needs to be operated during the use of the system for sending and receiving graphics.

Transmission System

Since the bandwidth of hand generated graphics is rather limited, it is possible to transmit this information in real time over a narrow band telephone line. It has been determined experimentally that the amplitude frequency relationship of handwriting on a blackboard is related as shown in Figure 3. It can be seen that for 0.1 inch peak to peak writing (practical limit of writing with chalk) the frequency is 14 Hz. Accordingly, 28 samples per second (Nyquist rate) are sufficient to completely describe the handwriting. In order to allow inexpensive and high quality reconstruction of the original writing a sampling rate of 40 samples per second was chosen. Since the X and Y coordinates are independent, 40 samples per second are transmitted for each coordinate. If the blackboard X and Y coordinates could be sampled simultaneously, sampling of the board at a rate of 40 samples per second would be sufficient.

*Available through Bell Telephone companies.

However, because of the time sequential sampling employed the delay between the X and Y samples would cause a spatial equivalent of phase distortion resulting in poor display quality. To minimize this delay the board is sampled 250 times per second, the output filtered by a sample and hold circuit, and resampled at the 40 Hz sampling rate for transmission.

The objective of using ordinary telephone grade transmission channels requires a very rugged transmission method. While nonuniform frequency response, phase distortion, poor signal-to-noise ration, and echoes are reasonably tolerable in voice communication, they can seriously affect the quality of visual information if analog transmission is employed. Since the Bell System offers a standard Frequency Shift Keying (FSK) data transmission system, it was decided to send the handwriting as digitally-encoded data, with a data set 202 type.

As mentioned, the blackboard has a very large resolution capability and to transmit information efficiently the resolution had to be arbitrarily limited. It was decided that 1000 by 1000 point resolution is more than sufficient for blackboard handwriting resulting in 20 bits per XY coordinate point.

Fig. 3 - Handwriting Frequency Response

Transmission is chalk-switched and half-duplex, analagous to voice switching in speakerphones. The system switches from the RECEIVE to the SEND mode when the chalk touches the blackboard. Lockout circuitry prevents writing from being transmitted in both directions simultaneously. Although data sets exist that allow simultaneous transmission in both directions, they are more expensive than half-duplex transmission systems. Furthermore, half-duplex transmission is necessary for multipoint conferencing so that any location may send to all other locations.

Experience has shown that half-duplex transmission is quite sufficient. People don't "fight" over the chalk the way they do over the microphone in voice-switched conference systems.

Display System

The display consists of a conventional television monitor and a scan converter-type television frame memory. The memory receives and stores the analog XY position information as charges deposited on the stored image is continuously scanned at standard television rates, generating a composite video signal compatible with commercial black and white or color television monitors. The memory switches between the WRITE and READ modes at a 15 kHz rate with writing occurring during the horizontal flyback period. This rate is sufficiently high to eliminate the need to buffer the incoming graphics signal. The 15kHz rate with a limiting resolution of 400 linear points per display width corresponds to a writing speed of 26 mseconds per display width (400/15000). This is much faster than handwriting which is unlikely to exceed 250 mseconds per blackboard width.

The storage device can be switched into the ERASE mode with a remote command providing a visible square erase spot for partial erasing. Full erasure occurs under the command of the remote CLEAR signal in 500 mseconds.

Although the system resolution is limited by the memory, the typical resolution of 300 by 300 continuous points is sufficient since the size of the television monitor limits the quantity of graphics that can be viewed conveniently. The storage time of the memory is such that a good quality display is maintained for at least 45 minutes of viewing. This is expected to be sufficient for most applications.

The display is a bright, high contrast white on black image that can be viewed in ordinary room light. For large groups of viewers several monitors can be interconnected or a large screen TV projector may be employed. The complete Blackboard System is shown in Figure 4.

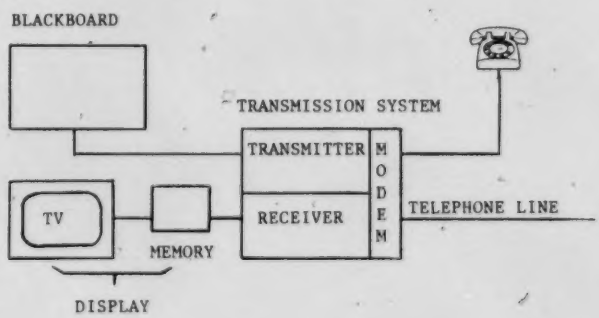

Fig. 4 - Blackboard System

Permanent Storage

A significant feature of the Electronic Blackboard system is the provision for recording the graphics transmission on a conventional audio tape recorder. Since the transmission is made over standard telephone lines, any suitable voice recorder may be used. If a two-track or stereo recorder is used the voice and graphics can be recorded simultaneously, providing synchronous playback. This means that lectures, training material, and conference notes can be played back at any time and the user will perceive them as they occurred originally. This contrasts with the use of "canned" material (except video tape) for which the audio is usally accompanied by filmstrips, viewgraphs, or other "still-type" graphics. In an educational application a remote student who is unable to attend the original presentation of a class session can make up the session at any later date simply by turning on the tape recorder and pressing a PLAY button on the student's local electronic blackboard system.

Because of transmission impairments introduced by telephone lines, recording directly from the telephone line would provide unreliable graphics storage. Therefore, in the electronic blackboard system, the tape recorder does not record directly from the telephone line. Rather, the incoming signal is demodulated, the jitter is reduced by a PLL (phase locked loop), and the signal is remodulated before it is recorded. (See Figure 5.)

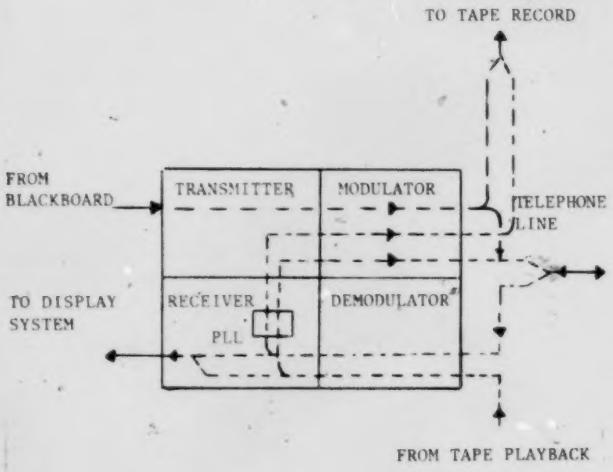

Fig. 5 - Tape Recording

During the playback the jitter and other impairments introduced by the tape recorder are similarly removed by the PLL prior to retransmitting the graphics over the telephone line. This reconditioning provides a very reliable recording and playback capability at local and remote locations and eliminates the need to send tapes by mail.

Where a hard copy of the graphics display is desired it is possible to connect a commerical copier to the television monitor's video output terminal. The copier can produce a good quality reproduction of what appears on the monitor in less than 20 seconds.

Conclusion

An earlier experimental electronic blackboard system has been used successfully at the University of Illinois for the past three years. That system lacked some of the features described here, such as partial erase and automatic switching between blackboards. Nevertheless, it has proved that graphics communication over voice-grade telephone lines is an effective way of providing remote visual communication in education. Electronic blackboards have been used experimentally at Bell Laboratories locations for remote conferences, sometimes tying together three locations in a multipoint conference. In response to a questionnaire distributed to the conferees, 80 percent of those attending indicated that the Electronic Blackboard was as effective as a visit to the originating location and that it eliminated the need to travel.

In the author's opinion there is evidence that graphics communication over the telephone network has utility in remote teaching and training and in business applications. It can reduce the necessity for travel to attend conferences or class sessions, and increase the size of the audience capable of attending a given conference or class session. The Electronic Blackboard system is but one solution for improving communication by adding realtime graphics to the voice transmission over ordinary telephone lines. Other systems are sure to follow.

Acknowledgment

The author wishes to acknowledge the significant contribution of the following in developing the Electronic Blackboard system: J. Kee, Jr., H. G. Mattes, C. B. McDowell, and L. E. O'Boyle, all of Bell Telephone Laboratories, Holmdel, News Jersey, and to thank J. W. Fitzwilliam, W. B. Cagle, and J. S. Courtney-Pratt for their continued advice and encouragement.

SLOW-SCAN TELEVISION
AN EXPERIMENT IN EDUCATIONAL TV
IN THE COPPER COUNTRY OF MICHIGAN

Jim Zygaj
Project Director

Slow-Scan Discussion

Slow scan is a form of video information transmission over communications channels with frequency bandwidths substantially less than those bandwidths normally required for video information. Typically, bandwidths of 4 to 5 MHz are required for video, but high resolution systems with more picture detail can require 5-10 MHz bandwidths. As a comparison, audio information takes a 3-15 KHz bandwidth. A slow-scan system can compress the wideband video to narrowband video and transmit this compressed video over an audio communications channel. A real advantage is that existing narrowband communications channels can be used once the video is compressed.

The reduction in bandwidth is done at the expense of transmission time. For example, a single frame of conventional TV type live video takes 1/30 second for transmission. If the bandwidth is reduced to 0.1% (5 MHz to 5KHz), transmission time increases by a factor of 1000. A single frame transmission time is then about 33 seconds. This is the required "scan time." During this time the video image must remain fixed. The scan time does not necessarily have to be fixed but can be varied depending on required resolution, frame size, and available bandwidth. Low resolution systems can use scan times under 10 seconds, while higher resolution systems can have scan times on the order of minutes.

Implementation

To implement a slow-scan system two major hardware items are required. At the transmission end, a scan converter or compressor is needed to convert the wideband video to narrowband video. This unit may require some form of image storage if the image does not remain fixed during scan.

Sampling of the video information is required for compression. At the receiving end, a video expander is needed to reverse the compression process. This unit must have image storage capability if it is to display a conventional TV image.

In addition, other hardware items may be needed. Synchronization has to be maintained between the transmitting and receiving ends. This may require that a sync signal be sent along with the compressed video signal. However, this is not necessary if a common sync signal can be obtained at both ends. A common signal is the 60 Hz power line frequency where both ends have the same power source.

A number of communication systems fail to pass DC, and this may require that some type of DC reference be established for the narrowband video. For example, phone line transformers will not pass DC. A DC reference must be established since the contrast levels are references to the same DC level. A method of retaining DC reference in a system is to FM modulate a signal carrier by the video and demodulate it at the receiving end. A FM modem for phone line use can be implemented easily.

Performance

As in any communication system, there are a number of factors which effect performance. For the video portion of the slow-scan system, the major parameters of performance are resolution and contrast. To be compatible with picture quality for standard B & W TV, the resolution should be at least 200 TV lines with resolution of 400 TV lines being desirable for more detailed images. Contrast as measured by the grey scale should be at least capable of five shades of grey with 7 to 10 shades being preferred.

The communications channel performance also has to be considered. The frequency bandwidth should be sufficient to pass the slow-scan narrowband video. Reduced bandwidths would degrade the video with the result being a loss of resolution and contrast. Contrast is related to signal amplitude. Noise or other causes of amplitude change will change the contrast. Of primary importance is transmission with a stable DC reference. Any changes in DC reference will change contrast.

Applications

The potential applications of slow scan are numerous. Slow scan can be used on a number of existing communications channels. Voice grade on equalized phone lines, narrowband video links, or direct closed circuit wire are a few. Phone line usage is possible for just about anyone assuming purchase of a compressor-expander. Radio links can be used to send pictures world-wide. Slow scan has been used to transmit the Mariner and Viking space probe pictures. FM subcarriers allows pictures to be broadcast with existing FM radio stations. Slow scan has been used in closed circuit monitoring of inner building facilities and parking lots. The applications for slow scan are many and varied.

Slow Scan at the CCISD

A slow-scan TV system was established at the CCISD primarily to determine its viability as an educational medium for the regional schools. The system uses the 100 KW local FM station (WGGL) SCA channel for broadcast to a service area in excess of 70 miles. The advantage with such a system is that widely separated rural schools can collectively obtain educational material and instruction that they individually could not obtain. A receiving site can be set up anywhere within the 70 mile radius for reception of slow scan. Use of an existing communications facility drastically reduced the cost of setting up a newer communications sytem.

The system was established for both simultaneous transmission of audio and slow-scan video. Two SCA channels, 41 KHZ for audio and 67 KHz for video, were required. The use of a 41 KHz SCA channel limited the slow scan air time since the 41 KHz channel was available only when the station was not in stereo. If the 67 KHz channel was time shared with audio and video, slow scan could have been broadcast all during the station's air time.

System Audio

The audio portion of the system consisted of several distinct parts. Normally, the audio program portion was recorded on the audio track of ½" video tape. To initiate scans in the video one second 25 Hz scan tones were inserted on the audio track when a scan was desired. This tone was inaudible in the received audio signal. This tone was detected by a tone sensor which triggered the video compressor. The audio output was sent to the FM transmitter over a 50-15KHz equalized phone line.

At the transmitter, additional equipment was required. To prevent overdriving the transmitter, the audio was sent to a limiter which limited the maximum signal level. The limited output drove an SCA generator in the 41KHz range. This signal was mixed with the 67KHz video and amplified before it was sent to the transmitter modulator.

To receive the signal a FM-SCA receiver (McMartin TR66) was used. This converted the 41KHz SCA signal to conventional audio. This audio signal was used to drive a TV monitor or CATV head end modulator.

System Video

The video portion of the system had the following configuration. The video portion of the program was recorded on ½" video tape with the use of B & W TV cameras. Graphics and slides were typical program material. The frame scan rate of one per ten seconds was fixed in the video compressor. Scans were initiated by the 25Hz tone previously mentioned.

The transmit unit video equipment consisted of the playback VTR, the video compressor (CVI 201A), a 8KHz low pass filter and a FM phone line modulator. The output of the Sony 3650 VTR was directly connected to the compressor input. The VTR was synchronized to the AC line frequency which was the source of the system sync. The output of the compressor was filtered to eliminate sampling noise. A FM modulator with a 10 to 15 KHz deviation was used to preserve the DC contrast information. A separate set of phone lines was used for the video.

At the receiving end of the phone lines, there were several equipment items in the FM transmitter site. The slow-scan video was demodulated by a FM demodulator (10-15KHz). The demodulator output was filtered to remove the carrier feedthrough by a 0-8l2Hz low pass filter. The filter output drove the SCA generator (67KHz) and was mixed and amplified with the 41KHz audio signal and the mixed output drove the FM transmitter.

At the receiving sites, 1 67 KHz FM SCA receiver similar to the 41KHz audio receiver was used to extract the slow-scan video. This receiver was modified to restore the DC reference to the slow-scan video. The receiver output was connected to the video expander (CVI 220A), where it was stored on a magnetic disk. Since the disk was driven by an AC sychronous motor, the disk was synchronized to the line frequency. The only adjustment related to sync was a fixed phasing to compensate for delays. The disk also had a separate sync track for storage of the video sync pulses. The 220A drove either a TV monitor or a cable modulator.

Technical Problems

A number of technical problems were encountered in the system. A major source of difficulty was in the 220A video expander. Several of the units were in remote unattended receiving site locations and had to run continously. If the unit was turned off, the phasing (vertical centering) had to be manually adjusted. After a period of operation (a few weeks to a few months) problems occurred. Two units failed catastrophically with damaged video heads. Several exhibited sync jitter or a complete loss of sync. Picture quality was observed to be degraded at times. Several times vertical centering was off. In addition, frame scans were intermittent or absent.

The catastrophic head failure, sync loss or jitter and picture degradation problems are attributed mainly to the continuous unit operation and the temperature changes at the unattended receiving sites. If the heads were slightly misadjusted, excessive wear would occur after a long period of operation. In addition, temperature changes could change the head/disk spacing causing further wear. It was found that even though the disk was enclosed, dirt accumulated on the disk and head. The source was probably particles from the disk and head contact. Dirty heads caused a loss of contrast in the picture quality.

Loss of vertical centering was attributed to two sources. Any temporary loss of AC line power (which happened occasionally) would cause a change in centering. In addition, if the sync jitter became excessive the centering could be shifted.

Another system problem area was the temperature induced shifts in the operating point of several units. It was noticed that the FM SCA receivers DC reference level (video receiver) would drift drastically by 0.5 to 1 volt. It was determined that the drift was not only receiver related, but that it also was common in part to all receivers. This was at the FM transmitter site equipment and was either in the phone line demodulator or the SCA generator.

Performance

The audio performance of the system was found to be acceptable except for two areas. At the WGCL site, as previously mentioned, noticeable crosstalk from FM main channel to SCA channel was present. It is objectionable primarily during periods of lulls in the normal program audio. The other area of degraded performance was in audio frequency response. The bandwidth was 6 to 8KHz and was restricted primarily by the bandwidth of the SCA channel.

The video performance was also degraded in several areas. The resolution as measured on a standard RETMA chart was found to be in the range of 200-250 TV lines. The major loss in resolution was determined to exist in the phone line modem and SCA generator and FM receiver. The slow-scan video bandwidth instead of being 8KHz was reduced to 3KHz with a consequent loss in resolution.

Technical Hardware Accomplishments

At the Holman School receiving site, a fixed location was established for the receiving site. Three classrooms were wired for closed circuit reception of programs. A scan filter was installed in the 220A video expander. This particular filter was a bandpass filter with a 750Hz center frequency used to extract the scan starting pulse train and minimize any noise that would cause intermittent starting.

At the transmitter, two units were designed and constructed for installation. An SCA mixer and amplifier was used to provide proper mixing of the audio and slow-scan video subcarriers. It also amplified the signals so that the injection level (typically 10%) could be measured. A 8KHz low pass filter was installed after the phone line demodulator to eliminate the carrier feed through.

Further Developments

At present, expansion of the existing system is not feasible. The cost and technical difficulties make expansion of the broadcasting system unlikely. It is possible to continue with the existing system. Until the cost of the expanders can be reduced and their performance upgraded to broadcast reliability, the system should not be expanded.

A future direction is the establishement of a distribution and VCR playback system. Slow-scan tapes should still be cheaper to produce than other wideband types. This system should have minimal technical problems since there is less hardware involved and typically commercial VCRs are reliable.

Conclusion

Slow scan from a technical standpoint has been shown to be feasible in a FM SCA broadcast mode. It also has been shown that there is a degradation in the video quality. Also the equipment is not of broadcast quality and it needs further improvement without becoming more expensive. With the continued development of large semiconductor memories (CCD's) the cost should come down for expanders substantially in the future.

Any future development of a FM SCA should consider the following areas. The technical aspects to the system will require substantial technical support. Problems are difficult to solve if air time is limited and there exists a viewing audience that requires minimal downtime. Receiving sites over a wide area must be made very reliable to keep servicing to a minimum. Moving disk memories should not be used.

CONTROLLED-SCAN TELEVIDEO: EFFICIENT UTILIZATION OF COMMUNICATIONS CHANNELS FOR TELECONFERENCING AND DATA ACCESS*

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Abstract

Audio-visual material is discussed in a generalized way which shows how transmission in real-, or quasi-real, time can be effected over a variety of channels, both broad-band, in cost-effective "sharing" modes, and narrow-band, such as telephone lines. This has special importance where the visual component does not involve motion (televideo rather than television).

Various conferencing functions may be seen to require only still-picture capability. It is shown that CSTV techniques are both sufficient and cost-effective for analogous teleconferencing applications including many educational situations. Reference is made to Northwestern University demonstrations of the sufficiency of CSTV techniques using only telephone lines in medical training, teleconferencing and off-campus continuing education courses as well as to other experiments elsewhere. Certain improvements, some perhaps crucial, are discussed.

In addition to the obvious cost-effectiveness of CSTV, certain advantages exist even vis-a-vis interactive CCTV when available. There is special application to the problem of sharing the resources of educational institutions including making data bases such as library materials accessible to students and scholars geographically remote from the source.

I. Information, Media and Their Manipulation

We may rely on a very intuitive view of information which tends to distinguish information into visual and aural categories. The former may be text (alphanumeric characters), graphics, mathematics, or pictures, including representations of three-dimensional objects. The latter includes voice communication, especially interactive speech, sounds and music. There is also the "audio-visual" combination. A variety of old and new technologies can provide information: books, correspondence courses, programmed texts, audio-cassette lectures, audio-cassette-with-slide lectures, sound motion pictures, broadcast radio and TV lectures, video tape presentations and, on the near horizon, videodiscs etc.

The leavening of a human being may be added to any of these in an INTERACTIVE RELATIONSHIP between the participants. The advent of computer-assisted instruction has filled a gap in the spectrum between a "canned" presentation of information and a traditional teaching situation but we must stress the element of INTERACTION as the important point. It is interaction between the student and the presentation of the information which transforms the transfer of information into the process of instruction.

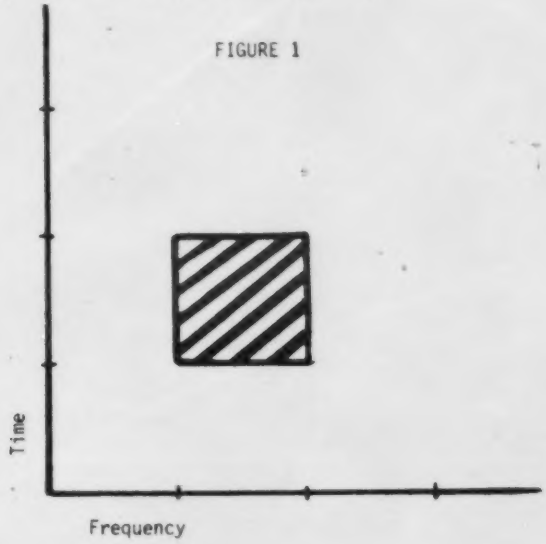
The popular use of the term "media" encompasses both passive systems and interactive ones and some distinctions should be drawn between them. Informational or recording media represent ways to store and present information to the student. Telecommunications media or telemedia involve transmission from one place to another and, for most of our discussion, we shall assume that the transmission is essentially instantaneous. Where we choose to relax this assumption, we shall distinguish between real-time and quasi-real time, the latter being transmission sufficiently soon after an expressed request for information that its receipt is timely and useful. Thus, for the purposes of our discussion, a telecommunications medium is a real-, or quasi-real-, time system which is basically interactive and it is the interactive feature which dominates and defines the distinction. Thus, a lecture broadcast to a mass audience which cannot immediately respond to or affect the course of the lecture might just as well be recorded and distributed, for the purposes of our distinction. The use of videotape in almost destroying the old distinction between "live" and "recorded" broadcast television is a case in point. To the passive watcher, it makes no difference. To carry our categorization a bit further, it clearly is the case that a teacher and a student on opposite ends of the same log is a completely interactive situation (always assuming the teacher has not cowed or "snowed" the student into passivity). A quasi-interactive situation is one in which the student or scholar has control over part of the presentation but cannot have it modified totally to fit his individual requirements. An example is the ability to select, delete, skip or conjoin materials such as might be the case with accessing of library resource-type materials. We can draw a parallel between real-time/interactive systems and quasi-real-time/quasi-interactive systems and we shall return later to the consideration of overnight "narrow-casting" of library materials to individual users. The distinction is not absolute, however, as exemplified by such techniques as computerized-teleconferencing whereby messages are transmitted in alpha-numeric form by teletype terminals and monitored by a supervisory computer. This is basically asynchronous with a participant's return to his terminal being real-time until he replies to all past messages and poses all the questions he has for others.

The nature of information and its manipulation becomes especially important when it must be transmitted over a channel whose capacity is limited. It is important for cost-effectiveness always to keep in mind that channel capacity is a resource which cannot be wasted without incurring economic expense. It has long been recognized that the world is facing a shortage of bandwidth in the frequency spectrum (which Marcel Thue, Chairman of the French Frequency Committee, has called a commodity "that should be treated as a permanent, but limited, natural resource like water"). New technology to provide additional channel capacity is no panacea. New methods or modes achieve cost-effectiveness benefits over prior modes only in certain regimes. Communica-

tions satellites, perhaps the most glamorous of the newer modes, become cost-effective vis-a-vis other modes for providing channel capacity only when the distances involved are in excess of 200 miles or where the terrain or access is especially difficult. Efficient use of any channel requires that redundancy of information be eliminated wherever possible. Looking at a stationary object for ten seconds over a full-TV link involves sending 300 identical frames of visual information. If it is possible to send only one frame in a form which is usable at the other end, 300 users could utilize the same channel in this way over the 10-second interval. Motion-television utilizes 30 frames per second to display motion but, on average, 95% of the information in one frame is redundant with that in the frames next to it. On a statistical basis, only about 5% of the information contained in a TV frame needs to be transmitted to depict even motion provided that the information contained in the preceding frame is available to be updated.

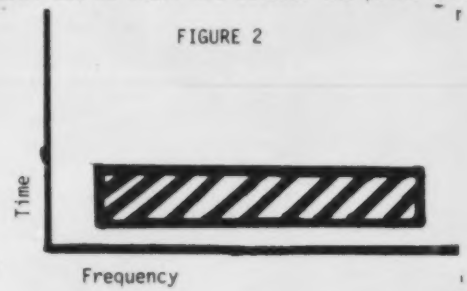
The TIME X BANDWIDTH product measures the information which must be sent over a channel. This may be represented graphically as the area on a plot of TIME (that a channel is utilized) versus BANDWIDTH (the frequencies utilized). (See Figure 1)

FIGURE 1



The first point is that this area should be reduced to a minimum since large values of the product will always be more costly than smaller values, that is, the redundant information should be deleted when possible. Thus, in our example, the information transmitted by TV is 300 X larger than it need be. Technically, further reductions are possible since individual frames internally contain redundant information in general but we shall not belabor the point here.

FIGURE 2



The second point is that technology can manipulate the form in which the information is sent. In Figure 2 we see that the same information can be sent in a shorter time provided that we utilize more of the channel bandwidth. This permits use of the channel in the time-division-multiplex (TDM) mode. For example, if the unit of information is one TV frame, each user can transmit one frame in 1/30 second by using the full bandwidth. We have glossed over some technical details such as how each user identifies which frames are "his" and how these frames are grabbed, stored and displayed. This is frequently referred to as a "frame snatch" or "frame-grabber" mode of operation.

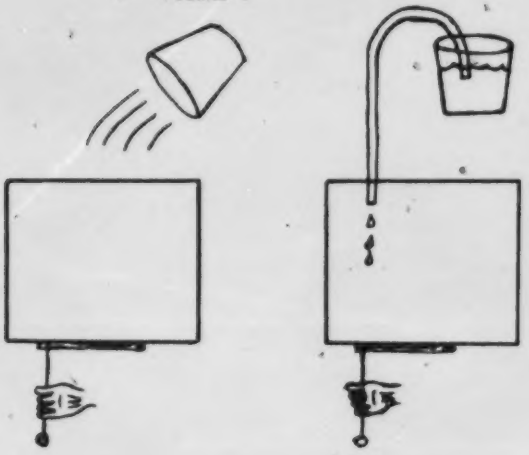
FIGURE 3

In Figure 3 we see that the same information can be sent over a narrower-band channel utilized for a longer period of time. Splitting a given channel up into narrower frequency bands is use of the channel in the frequency-division-multiplex (FDM) mode. Here again, if we use the example of one TV frame, a user can transmit one frame in 1 second by using 1/30 of the bandwidth or in 33 seconds by using (approximately) 1/1000 of the bandwidth. This ability to adjust the rate of transmission of a frame to accommodate the channel is what we call the "controlled-scan televideo" (CSTV) technique. When we restrict the speed to a slow rate the technique also goes by the name of "slow-scan TV" (SSTV). Since there is approximately a factor of 1000 between the bandwidths of television channels and audio channels such as telephone lines, we see that an ordinary telephone line suffices to transmit a frame in about 33 seconds.

We note that the technical detail here is the ability to send the information in pieces over the time

the channel is employed, accumulate the pieces, reconstitute the original information and display it in the original form. We wish to emphasize that the systems capability required, once the information is stored, to maintain and display the information in its original form is the same as that in the frame-snatch mode. The front-ends are different and may be epitomized by Figure 4 where the reservoir (frame memory)

FIGURE 4



is filled either by dumping a bucket of water (frame-snatch mode utilizing a broad-band channel for a short period of time) or by siphoning the water through a narrow pipe (slow-scan mode utilizing a narrow-band channel for a longer period of time). The rest of the technology is essentially identical.

There is complete interchangeability between all forms of information: they can all be encoded into electronic form. While this is especially obvious when considered in terms of digital encoding, it is also true for analog techniques. The following list shows the actual implementation of two different forms of information in terms of different configurations (with the same time-bandwidth product):

INFORMATION	TIME	X	BANDWIDTH
1 Video frame:	1/30 second	X	(TV Channel)
	60 second	X	(Teleph.Channel)
	6 second	X	(SCA FM Channel)
16 seconds of audio:	16 second	X	(Teleph.Channel)
	1/30 second	X	(TV Channel)

II. Teleconferencing Mode

Before we can address the utility of these ideas in the general area of teleconferencing we must recognize that there are categories of conferencing function which should, perhaps, be recognized lest we fail in a vain attempt to be all things to all people. We shall arbitrarily divide the functioning of conferences into three modes, which we shall call "executive", "presentation", and "working session". This is certainly somewhat simplistic and arbitrary, but it may have some utility in discussion. It should also be noted that our categorization has a parallel in the hierarchical structure of many organizations.

Executive This category derives its name from the fact that high-level executive interactions are frequently of this type. It usually involves a small number of persons; frequently 3-5 people, sometimes two with staff backup. Personal contact is the goal, being an affective interaction rather than an informational one. Visual aids are not usually employed but "body language", facial expressions, and vocal inflections are important. It is frequently the case that communication of credibility is more important than the communication of information. Job interviews are frequently of this type as are initial meetings of people who have not met before.

Presentation This category of conference usually involves one person addressing n persons who can ask questions and make comments. The person "having the floor" uses visual aids: overhead projector, viewgraphs, slides, charts, etc. Color is frequently recommended for impact, even if not necessary for delineation of information. (Some of the educational literature warns of the diversionary effects of color for its own sake. One must be constantly aware of the danger of "expressing the speaker's creativity rather than being concerned exclusively with the impact on the audience"). Visual information is essentially one-way (1 to n) with audio interaction fully bilateral. Visuals need not have resolution sufficient to read a full printed page. In fact, visuals for presentations should not be too complex nor should there be too many. Neither should they change too rapidly. Usually, "presentations" require preparation.

Working Session This is always fully interactive and usually democratically so. It occurs among peers and tends to be informal. The people involved know each other to some extent, frequently very well. The number of persons involved may be as few as two or as many as 20. (Recalling the old saw that a "camel is a racehorse designed by a committee" reminds us that there are practical limits to the number of people who can profitably work together in a fully interactive manner.) Visuals are important (pictures, graphics, equations, numbers, computer printouts, text, documents, instruments or objects, etc.) and are fully multilateral. That is, the visuals may originate with any of the participants. Sometimes a working session must result in a written draft or recorded consensus. The ability to read a printed page is sometimes important here.

There are several points which are relevant here. First, we are interested in the extent to which teleconferencing, especially of the cost-effective sort we are most concerned with, can substitute for travel to effectuate in-person meetings. The limitations of a still-picture system are least noticeable when the contact is predominantly informational or cognitive rather than affective. We would, therefore, expect that the presentation and working-session type conferences lend themselves most readily to CSTV techniques. Our experience in several teleconferencing experiments bears out this expectation as to the adequacy and effectiveness in these categories. Certain attention to detail and important modifications are necessary to produce these empirical results which are more sanguine even than others reported[1] using similar equipment.

The situation is less clear with respect to executive-type conferences and we have little experience with assessing the limitations of the technique in situations where one is often concerned with evaluating the person making a presentation as much as the detail of the presentation. Naively, one might expect that personal contact is absolutely required and that only a broadband, motion-television system could suffice to

simulate the personal interview situation as closely as possible. One might want full-color motion-television to "see the countenance pale, the hands tremble, and the beads of sweat form". On the other hand, we shall at least mention later some considerations which might militate in favor of a more limited system having actual advantages. Considerations of possible advantage aside, let us keep in mind that executive-type conferencing is perceived to be more important as one looks higher in an organizational hierarchy. While it is true that higher-level management tends to travel more per capita for the purpose of having meetings, it is also the case that most organizations have many more indians than chiefs and the major part of the travel occurring in an organization is done by the scientists, engineers and middle managers who are concerned with the actual work on projects rather than by the high-level management who are more concerned with making the decisions as to resource allocation in different areas. Perhaps the most important caveat to be kept in mind is that a teleconferencing capability such as we are concerned with constitutes a supplement to personal travel. There are trips which are foregone because of inconvenience and expense and there are interactions which are never made which might have resulted in some productive work. We thus must add the "lost opportunity" cost to the other costs of travel although it will always be difficult to quantify this consideration.

III. A Model for Education

We consider the model of a professor and a class of about 25 students to be a good one for our discussion. We believe that the essence of the situation is the interactive relationship between the teacher and the students and, we should not forget, among the students. We can easily argue for a set of criteria. We wish to provide to students who are off-campus the essence of what has traditionally been available on campus. Further, we wish to do this in ways which do not require large perturbations in the procedures familiar to both professors and students. The essence certainly includes the ability to interact; this means that two-way voice contact is an absolute necessity (including the ability of the students to communicate with each other). We take as our model of the classroom one in which the professor communicates via voice and an overhead projector. Harking back to the blackboard is too retrograde a step! Scientific conferences have almost totally eschewed the blackboard in favor of the overhead projector (and slides) and the trend is the same in the classroom. One has at least part of the armamentarium of visual aids available today: the ability to write in real time (as with a blackboard) but also the ability to pre-prepare material, use typewritten text, graphical aids, and pictures.

We also make two assumptions. Using our model, one is quite good: motion is not needed in the classroom for presenting information. The second is more arguable: motion is not needed to provide whatever component of affective interaction between professor and student may be important in creating the process of instruction. The literature[2] (e.g. see Travers (1967)) appears to bear this out. Some have even gone so far as to observe that motion may be more often a distraction than a help. The assumption is certainly more plausible in the situation of advanced university-level courses and mature, continuing education students. This assumption appears validated in work done at Northwestern University both with Northwestern resident students and with graduate engineers employed at an industrial location some 25 miles from campus. This work is described here and elsewhere [3]

We believe, in general, that information transfer alone, together with the necessary intellectual interaction to make instruction occur, does not require motion. Motion is usually required only when it is desired to elicit an emotional response. In the classroom, it is only those rare, peculiarly animated and stimulating professors with a unique teaching style who perform much of their job requiring motion. Even here, it is perhaps the case that the "extra" provided by these, often popular professors, is more in the sphere of entertainment than instruction. For the vast majority of cases, one is as often likely to observe a professor's "tic" as his animation and, so, the ability to see the professor "move" may be obviated.

We are now at the point of defining a system to provide off-campus education where the instructor is temporally but not spatially present. It must be interactive with two-way voice (as in a classroom). It should provide the ability to display visuals, both pre-prepared and constructed in real-time in response to questions or a perceived need (as in a classroom with an overhead projector). It need not, however, display motion (again, as in most classrooms). It should be available wherever the students are and, ideally, it should be possible to teach students who are not all in one place. Further, to help educational "productivity", it would be desirable to load on-campus classes incrementally with students from their off-campus locations. We have in mind that the total of students, both on-and off-campus, would be less than 50 and, preferably, about 25. This is to foster interaction which declines with both too few and too many students.

At Northwestern University, we have experimented with teaching via SSTV using primarily RCA Videovoice equipment and these experiments are more thoroughly described elsewhere[3]. Likewise, the equipment currently available is describe elsewhere[4] but we may summarize the system used. RCA Videovoice provides a transceiver capability at each terminal with full TV resolution possible (nominally 525 lines but actually about 340 lines, as is the case for home TV) with a transmission time of 60 seconds. Another option is a "freeze-frame" mode, which stops any movement much as a snapshot does and which has half the vertical resolution but may be transmitted in 30 seconds. During a transmission, the screen at both the transmitter end and the receiver end goes blank. This is an unfortunate feature of the system design which is a function of RCA's use of a scan-converter storage tube as the memory element which cannot both read and write at the same time. Colorado Video's equipment, based on a video disc, has somewhat inferior resolution and transmission-time features but "paints" the new picture over the old one in vertical strips moving from left to right. The LANTEL (Los Alamos-Northwestern TELevideo) System prototype paints the new picture on top of the old one in horizontal stripes from the top down. The blanking feature of the RCA system was modified by us in a later experiment in a way which should be done in more ideal systems.

The gray scale and resolutions were compatible with usual home TV standards and were more than adequate for the purposes of teaching courses in physics and chemical engineering. Our standard mode quickly became the 30-second, frozen-frame, half-resolution transmission. Lower resolution yet and two-level gray scale (instead of the full gray scale of about 16 levels) would have sufficed for most of our classroom transmissions of a mathematical nature.

The first concern was how to bridge the 30-second "gap". This was easily done by "talking" and making a special effort to "interact" with the students during the deadtime. This was, of course, possible because of our use of two telephone lines for all these tests: one for audio, the other for video. This is, in fact, not necessary since the video channel is not in use much of the time. It is technically possible to "time share" the audio channel for the occasional video uses in such a way that the student is not aware of the "gaps". This has not yet been implemented in an SSTV system but the technology is straightforward, if non-trivial, to provide this "voice-over-video" capability for one telephone line. This is part of our ultimate desideratum of a one-unit-per-person system for students in their individual homes.

During the transmission time, discussion was continued about the picture which had been displayed and was initiated about the picture in the process of being transmitted.

Despite this straightforward solution, which was operationally satisfactory during our on-campus experiments, there is a technical one which was implemented during a four-week experiment in which a course in Human Factors Engineering was taught during 12 lunch hours by a professor at Northwestern in Evanston, Illinois to students at Motorola's Research Center in Schaumburg, Illinois, about 25 miles away. During the on-campus experiments, we determined that the 30-second delay in the visual transmission was acceptable to the student if the teacher accommodated to the medium by using verbal discussion to finish a previous point or to prepare the student for the next transmitted visual. Nevertheless, students indicated that they would like to see the last frame of video retained. That is, there is merit to having both the current picture and the one just previous. This advance in the system technology requires additional memory. Because we project memory to become increasingly economical in the future, we decided to add this feature for the Motorola demonstration on an ad hoc basis. We added additional memory at the receiver end in the form of an Hitachi Disc with appropriate switching.

The system at the remote classroom receiver now had two display monitors, one for "current video" and one for "previous frame". It is possible to operate the memories and displays as follows: at initiation of a new video transmission, both displays are switched to observe the most recent previous frame (for the RCA system, this obviates the annoyance of the screen going blank); upon receipt of the full frame into the primary memory, the primary display is switched to observe the new frame upon teacher command or when the frame is complete.

There are several advantages to this specification. Most important, the teacher is able to present lecture material without the student necessarily realizing the existence of a finite video transmission time. All that is required is that the teacher know what is to be transmitted 30 seconds before it is to be shown. The transmission occurs sub rosa and the new frame "appears" instantly (in 1/30 second). The only circumstance where the student would notice the delay is in response to a specific question or in a clumsy teacher presentation. In any case, the acceptability of the delay has been established. An additional advantage, which came out during our field test, was the instructor's ability to transmit two video frames, when desired, and talk about both simultaneously (although they were actually transmitted sequentially).

One need came to the fore almost immediately in actual usage: the absolute requirement for quality audio. Since the prime mode of interacting is aural, it is important that audio quality be maximized. This is true both for the student's ability to hear the professor and the professor's ability to hear the students (and the student's ability to hear each other if they are in different locations). We quickly adopted the use of lavalier microphones for the professor, Bell System Model 50A Portable Conference Sets for both studio- and remote-classrooms, and voice-actuated microphones with constant-level controls.

Concern about the need for a professor to make great perturbations in his teaching style in accommodating to the SSTV format was alleviated during the Motorola test. The instructor was a brilliant teacher but one who held no special brief for this sort of media teaching. Accordingly, he pushed the medium very hard with little attempt at accommodation to its possible limitations. Lectures were not carefully prepared, he used figures from thick books, scribbled on cards during the lecture for transmission to response to questions (and, indeed, as part of the main lecture) and mixed drawings of different sizes and aspect ratios. At times, the lectures were mostly aural, necessitating pictorials of the professor rather than course-related material to "fill the gaps". The student-camera-operator was more than equal to the task (using a zoom lens on the TV camera) and both the medium and the last-frame feature came through the test successfully.

Even before the last lecture, the Training Director of Motorola "accepted" the medium. That is, Motorola was willing to pay tuition for degree programs if Northwestern's Technological Institute could deliver the course material largely by SSTV. During the last period, a crude student "evaluation" was done. Most significant is the following:

1) More than 80% of the students felt that they had learned as much in the SSTV lectures as they would have in the same time spent in a live lecture.

2) 90% strongly preferred to be able to stay in the plant to take courses.

3) Over 70% thought the last-frame feature was an important addition.

4) 90% of the class was satisfied with the tonal quality of the telephone audio.

5) 90% said that the video picture helped to direct and focus their attention for enhanced learning even when the picture was not absolutely essential for the presentation. Video tapes of the remote classroom situation show students asking questions of the (still) monitor as if it were the professor (even when the monitor was showing a still picture of something other than the professor).

6) 60% would pay tuition money themselves to receive courses by this medium.

7) The members of the class were split as to whether they thought motion was necessary. We think this was not due to the demands of the course but, rather, to the erroneous preconception of the class that they would be seeing CCTV!

8) The class was not entirely comfortable with SSTV. About half said they "didn't like it" but couldn't pinpoint why.

We think the individual student evaluations supported the medium. In the design of this minicourse,

the students never met the instructor in person; they were not prepared for the medium (most expecting CCTV); they participated voluntarily, on their own time, and without formal course credit. Perhaps most significant of all, almost every student stayed with the class sessions from the first to the last one.

We should stress that courses taught by any medium should be carefully prepared and should utilize whatever is possible in the way of prepared material to be handed out in advance of the class: lecture notes, books, problems, pictures, etc. It is the "real-time"aspect of classes which the medium provides. Media teaching points out most forcefully the stupidity of much classroom technique, wherein teachers copy material from their notes onto a blackboard from which students dutifully copy the material into their notebooks without too much thought intervening. Media teaching points up the fact that classroom time is precious and should not be spent on secretarial functions. Media teaching of this sort is good experience for any teacher in improving and understanding classroom technique. Compared with ITV, SSTV has an additional advantage which came out during interviews with individual faculty members who were exposed to the system. Many faculty people are not partial to TV teaching because they feel exposed and ill-at-ease before the camera's all-seeing eye. SSTV severely delimits this "invasion of the classroom" and should be easier for many faculty members to adjust to than an ordinary CCTV.

IV. Features of CSTV - descriptive and normative

In our previous description we are aiming at a system which can use the ordinary telephone network to transmit still pictures and two-way voice. Television pictures are familiar and acceptable to almost everyone and suggest the television set as the display of choice. Telephones and television sets are essentially ubiquitous and the existing 1/8 Billion telephones in the U.S. are in place, reliable and interconnectible. If all our transmissions, both audio and visual, can be sent over ordinary telephone lines, we recognize that conference-call features, high fidelity gear and audio recording equipment already available at consumer-type prices become instantly available to a narrow-band audio-visual telecommunications system. We have reviewed elsewhere various techniques for doing this.

We believe that the system which has the most general applicability, and which will benefit from economies of scale resulting from applications in many other areas, is CSTV. Here, the input instrument is a TV camera but the transmission medium may be the telephone.

We can review again the simple argument which suffices to show that televideo can be used over ordinary telephone lines. A communications channel is able to carry information in much the same way that a pipe is able to carry water. The carrying capacity of a communications channel is measured by its "bandwidth": the range of frequencies which it can transmit without substantial distortion and loss. The unit of frequency is the "hertz" which is equivalent to the older unit, "cycles per second". A voice-grade telephone circuit has very approximately a bandwidth of about 4.5 KHz (where we denote "thousands of hertz" by "KHz"). That is, it carries only audio frequencies. We may call this a narrow-band channel. An FM-radio channel, similarly, is also "narrow", being about 15KHz wide. A black-and-white TV channel, on the other hand, transmits 30 frames per second and requires a bandwidth of about 4.5 MHz ("millions of hertz"). This may be called a "broadband channel". Speaking roughly, a full television picture (with motion) requires the equivalent of about 1000 voice-grade telephone channels. Bell's Picturephone is no help in this regard, being a low-resolution television system which presently requires about 300 voice-grade equivalents. We stress the well-known fact that Bell's Picturephone cannot be transmitted over ordinary telephone circuits, since it is intrinsically a 1-MHz system. Suppose, however, that we do not send 30 frames per second but, rather, only one new picture in 33 seconds. We thus see that the information rate goes down by the product of 30X33, or almost 1000. This lower rate can, indeed, be accommodated by an ordinary voice-grade telephone circuit. All we have given up is the ability to transmit motion and the ability to change pictures more often than once every 33 seconds.

We should mention the possibility of redundancy-elimination as discussed earlier. Because of the large redundancy of information between most TV frames, it is conjectured that Bell's Picturephone might be "squeezed down" to fit 10-20 voice-grade telephone lines eventually. Likewise, such projects as GE's Sampledot System promise to provide some compression of bandwidth for ordinary telvision. Present techniques have indicated approximately a factor of 4 compression due to frame-to-frame redundancy and another factor of two compression due to intra-frame redundancy. Interestingly enough, the hardware required for redundancy elimination is very similar to the configuration required for CSTV and the techniques may all be expected to converge. The point now, however, is that we recognize the frame-to-frame redundancy and eliminate it entirely by eschewing motion. The intra-frame redundancy can be eliminated as well with televideo as with television (actually more easily).

The issue of resolution is a more important one. This refers to the ability of a picture to discern and distinguish detail. Very roughly we may speak of the number of picture elements (PELS) contained in a picture. Each picture element may be assigned a measure of its color or, in the case of black-and-white, its gray scale. This permits the rendition of a picture, scene or object which has shading in much the same way that a photograph does. The number of TV lines is roughly proportional to the square root of the number of associated pels. Thus, a TV system with 500 line resolution has 4 X the number of pels (and, hence, four times the information) contained in a picture with 250 line resolution. American NTSC television has of the order of 500 (nominal) line resolution. It contains about 1/4 the resolution needed to read a page of typescript or a page from a printed book (which requires about 1000-1200 line resolution). Bell's Picturephone is DOWN by another factor of four from NTSC standards. That is, Picturephone pictures contain about 1/4 the information of a standard TV picture and about 1/16 of what is required to read text.

When we discuss the fact that a still-frame of video can be transmitted over a telephone line in about 33 seconds, we are referring to a new, standard American TV frame. Actually, we can get by in most circumstances with a picture whose resolution may be less than that of a TV picture and this lower-resolution picture may be transmitted proportionately faster. For example, a still picture of Picturephone-type resolution could be transmitted in about 8 seconds. On the other hand, a picture which we should like to be of higher resolution than ordinary TV can be transmitted in proportionately more time. This actually provides a flexibility which an ordinary television channel does not have.

We may ask why this technique has not already been used widely since people have been trying to squeeze pictures through telephone channels since the beginning of the telephone and telegraph. Variations of the basic idea have, in fact, been in use. However, some new technological developments have made this technique feasible and attractive only within the recent past. These pertain both to the wide acceptance of the television display and the development of "memory systems" for storing the large amount of information contained in a picture. Whereas in a television picture, the information is essentially stored in the channel, in that it is used as fast as it is transmitted, the "controlled-scan" part of the description implies that the picture is sent piece-by-piece over the narrow-band channel (at whatever rate is characteristic of that channel) which means that the picture must be stored at the receiver end and, when the picture is complete, displayed. We can carry further our analogy of Figure 4 to filling a water reservoir with a trickle of water from a thin hose. When the reservoir is full, the water may all be dumped out at a high rate. At the receiver end, the memory unit must serve several purposes: it must accept the picture information at a "slow" rate appropriate to the telephone line; it must accumulate and store this information in the proper sequence until the picture is complete; it must then spew the information out rapidly and repetitively (at video rates) to "refresh" a television set so as to maintain a flicker-free (but stationary) picture on the TV set. Memories which have been used to this end include a slowly-decaying-phosphor CRT (as on a radar screen), magnetic video discs, scan converter storage tubes, and digital computer memories. It is now possible to build a "box", about the size of a breadbox, which connects an ordinary telephone to an ordinary TV set and which contains no moving or glass parts (i.e. completely "solid-state"). At the other end, we have a "black box" which connects a TV camera to the telephone. The ever-decreasing cost and spatial volume of memory makes more capability possible. The transmission of a completely new frame takes 33 seconds over a telephone line. However, with multiple memory capability and some electronic sophistication, it is possible to "update" a picture in considerably less time. To move a pointer in real time or make underlinings, circles etc. would take only a second or two. Writing could occur essentially in real time if that were deemed to be important.

The "appearance" of instantaneous transmission of still-pictures can be effected by the use of multiple memories and the "foresight" to plan the transmission of a picture 60 seconds in advance of its complete receipt. From the vantage point of students, the CSTV system in teaching has much the appearance of CCTV. From the point of view of the instructor, however, CSTV is much easier to use, lending itself quite readily to his usual style and mode of teaching. This obviates a salient cost of media teaching: the development of new software. All the flexibility of an overhead projector is available with the additional convenience of being able to point an on-line camera at an object, person, text or picture and display it to his students. Another salient feature of CSTV is the flexibility to vary the resolution of the visuals employed. It is possible to display higher resolution than ordinary CCTV or broadcast TV permits. Certain special modes of operation permit either the use of high resolution monitors to read text or the use of ordinary TV monitors to do the same. The CSTV communications system is available wherever there is a telephone and the whole transmission, both visual and audio, can be recorded on an inexpensive audio cassette. The universality of application of these techniques in many fields augurs well for continuing reduction in cost as economies of scale take hold.

V. Remote Library Service and Other Data Access

Libraries remain our main means of storing information in forms which can be conveniently retrieved by scholars. The scope of library service has been expanded to include audio-visual materials in addition to the standard fare of books, journals, reports, and written/printed documents of all sorts. We shall consider the concept of library service to encompass any and all types of data. We have seen that they can all be expressed, stored and delivered in the same way.

There are two issues which are important. First is the need for inter-institutional sharing of these expensive resources. Second is the need for scholars (researchers, students, etc.), who are not geographically near to a library, to have as full access as possible to the materials readily available to scholars who are local. There is the "democratic" goal of providing equal (or, as equal as possible) access to scholarly materials by persons wherever they may be. There is also the practical circumstance that the scholarly community cannot afford the economic cost of too much redundancy. It is becoming increasingly difficult for all libraries to maintain complete collections of journals and the problem is especially acute with respect to specialized materials needed by only a few scholars.

There is the concomitant problem of fashioning a mechanism for providing dissemination of scholarly information which, by its specialized nature, does not justify even the scale required by ordinary publishing modalities. The "preprint system" and the "invisible college" of scholars in the same narrow speciality are ad hoc responses to this need but this mechanism is highly undemocratic and too unstructured to be acceptable as a solution to the problem. We here shall describe a system whereby telecommunications can be used to share institutional resources, to make all the resources of any and all libraries available to any scholar no matter what his/her location and to provide a means for "micropublishing" information which will avoid the inappropriate costs of the usual print media and yet will maintain the accessibility needed. On the U.S. scene, there are additional problems with customary modes of operation which impend because of changes in the U.S. copyright laws. We shall describe a system which certainly does not circumvent the law but which provides a mechanism for accommodating to it in a way which may be acceptable to libraries, scholars and publishers/authors.

We envisage a tiered system with primary libraries, secondary and, perhaps, tertiary libraries, etc. and, finally, the ultimate user in his/her office. The primary libraries would consist of national resource centers such as the Library of Congress and major collections, especially those with unique materials. The secondary and tertiary libraries would have significant collections of their own (less so as we proceed down the hierarchy) but which would most importantly serve their local communities as secondary and tertiary distribution points for library materials. The individual scholar would communicate either to a secondary/tertiary center or, possibly, directly to the primary source. The specific route would be determined solely by economics, availability of equipment and communications channels and the need for speed.

Let us review the present and currently-projected situation to see why any changes are needed. We have developed, and are continuing to develop, modern and

powerful methods for doing bibliographic searching of various data bases. However, once the needed material is identified (and this can be done within hours and, utilizing only the telephone, from essentially any part of the U.S.), there is no follow-up document delivery capability which is comparable. Ben Jacobson of the Northwestern University Library has referred to "20th century search procedures and medieval distribution methods".

The usual procedure involves several steps:

1) Reference interview with a professional librarian

2) Use of a computerized data base to determine the references which one thinks one needs (including materials which reading of the abstract, table of contents or a few pages would eliminate from consideration)

3) Determination of which library or data center has the desired materials

4) Ascertaining if the library in 3) has the material available

5) Obtaining the material by interlibrary loan of the original material, photocopy or microfiche copy

6) Awaiting physical delivery, usually by the U.S. Postal Service (or worse).

Step (1) is done reasonably well with present-day methods. For people remote from the librarian, the telephone works reasonably well but could certainly benefit from a visual component, especially in technical fields. Here a teleconferencing capability to supplement the usual telephone would be of value.

Step (2) is done superbly well at the present time and the situation continues to improve. This step frequently involves only minutes or hours to accomplish. However, the material identified by Step (2) is frequently not useful and this fact can be ascertained by skimming or browsing through the document.

Step (3) is presently done in ways which bear improvement. The reference librarian or one's own judgment can usually provide good input as to likely repositories of specific identified documents. The data bases could certainly be modified (albeit at some not inconsiderable expense) to include this information.

Step (4) is done well only by sophisticated libraries. It has been estimated that considerably less than a majority of the libraries in the U.S. can tell conveniently (without looking for the document in the place it is supposed to be) whether an item in their catalogued collection is available at any given time. Some difficulties afflict even the most sophisticated libraries. The document may be signed out to a borrower, at the bindery, stolen, misshelved or in process. There is, in fact, a distinct problem involved with the increasing power of the search procedures and the increasing mismatch with the follow-up. There is the danger of increased frustration due to enhanced expectation without commensurate ability to deliver. The use of interlibrary loans of original materials is actually counter-productive in that useful materials (identified by increasingly-many users with access to search procedures) will be unavailable to all users after the first one has removed them from circulation. The clear answer to the difficulties involved in Step (4) is to make the primary library a "closed-stack" system with the original materials not circulating beyond the direct control of the professional library staff. At one stroke, all the factors militating against availability of catalogued material would vanish. Of course, we must provide alternative means for users to access these materials.

Step (5), the interlibrary loan problem (described above) aside, is very labor-intensive. At present, some person must physically remove the document desired from its repository and make a copy of it. This first step cannot be eliminated by any system, given that the primary source material is initially in traditional form. However, we can hope to eliminate the need to duplicate this wasteful and time-consuming operation a second, third, etc. time. The making of microfiche copies already has this convenient feature. We should at least like to configure a system which does not need to return to "square one" each time a document is called for. It would further be most useful if the second (and subsequent) call-up were done in a machine-accessible, if not machine-readable, mode which lends itself to automation. Not only would this computer-accessed mode permit labor economies; it would also provide a means of accommodating to the copyright laws (and preventing the present system of inter-library loan and photocopying from collapsing entirely) by permitting accurate bookkeeping of accesses (and payment of appropriate royalties to publishers)for full-text transfers while providing open access to needed materials by users.

Step (6) is presently such an abomination that it bears little comment. The material must be packaged before being committed to the not-so-tender ministrations of the Postal Service and the delays and expense are considerable. We are at least solaced by the fact that the U.S. Postal Service usually produces delays and damage only; foreign services in Less Developed Countries have a very good record of losing the materials entirely.

Some Concepts We should distinguish certain ideas which bear emphasis. We refer to machine-accessible as contrasted with machine-readable. Even for the future, not all materials will ever be machine-readable although it is possible that text-only journals could be provided in this form. The enormous body of existing literature cannot be attacked from the machine readable point of view and pictures, complicated graphs and audio-visual materials will always be difficult to store and retrieve in a machine-readable fashion. Machine-accessible, on the other hand, refers to the ability of a computer to locate and access stored information. We suggest that a general scheme, which could be implemented at the present time, is to consider a videofile (either on 2" videotape, magnetic discs, videodiscs of whatever sort achieves success in the market-place, etc.) as the primary storage medium (after the first access of the material by a user). We have seen earlier that all kinds of information can be stored and transmitted in electronic form and that the storage can be made increasingly dense with more sophistication in redundancy elimination techniques. Most important, the video frames (of written material, audio, etc.) can be accessed by computer control on an individual frame basis. We must, of course, provide the flexibility to store frames of sufficient resolution to encompass all the material we are likely to be interested in.

We must distinguish real-time and quasi-real-time. It is frequently not necessary to deliver documents or materials in real time, which is arbitrarily defined to be receipt within minutes of request. Quasi-real-

time refers to delivery sufficiently soon after an expressed request for information that its receipt is timely and useful. As an example, we submit that days or weeks is often too long but that overnight delivery often would constitute quasi-real-time response. This distinction is very important because cost-effectiveness considerations militate in favor of using whatever communications channels are available, especially those which are under-utilized and cheap. At present, the vast telephone network of the U.S. is fantastically underutilized between the hours of 2 a.m. and 8 a.m. (approximately, since the time zones create some overlap at the ends). This is an available resource which should be employed even at existing rates. Another issue, of course, is whether the general good would not be well-served by having the telephone long-distance tariffs break at an additional point. They presently change at 8 a.m., 5 p.m., and 11 p.m. and we would suggest that another break, at 2 a.m. (or some time to be determined), might be in the interests of the nation and the telephone companies.

Another concept which is important for the issue of document delivery is that of resolution. We are considering video frames of flexible resolution and gray-scale (even color) capability. PROJECT INTREX,[5] among others, has identified 1200-2000 TV line resolutions as adequate for almost all library materials. We must provide the ability to store materials at this resolution and the ability to display them or, alternatively, reconstitute them at this resolution. When we deal with text only, there is another alternative as well.

There is actually no reason to reproduce exactly the format of present book technology when we enter the electronic age. Certainly there is nothing sacred about the need to turn pages although, should studies bear out that the tactile sensation is important, technology can provide that as well. In particular, the need for 1200 line resolution is predicated on the fact that most books involve text of about 80 characters per line. A TV display of American (NTSC) standards can display only about 40 characters per line. Interestingly enough, studies have indicated that 40 characters per line is more acceptable to readers than is the 80-standard. The data do not immediately scale to TV displays, of course, but Paterson and Tinker and Sir Cyril Burt have published studies[6] of legibility tests wherein it appears that 40 characters per line is optimal. Whether these tests carry over or not, it is clearly the case that newspaper compositors use the rule-of-thumb that one-and-one-half alphabets is what is best for newspaper columns. These raise not only the possibility of reformatting texts to utilize 40 characters per line but the possible advantage of so doing. It is straightforward to reconfigure standard text to this format and constitute an infinite scroll, 40 characters wide. This technique, by the way, indicates that the telephone line is more than adequate to keep up with the capability of anyone but the most expert speed reader.

Configuration of a System We shall describe in what follows an example of how Controlled-Scan Tele-Video techniques can be applied to employ these concepts to provide a library distribution system which can provide a good match to the sophisticated search techniques presently deployed. We envisage the telephone system to be the primary means of interconnecting primary, secondary, tertiary, etc. libraries with each other and with users. We adopt several principles as design philosophy. Among these are the following:

1) Use the telephone system.

2) Use CRT displays as either the primary display mechanism or as an alternative to hard-copy. Standard TV sets would be especially desirable.

3) Utilize existing technology wherever possible. Especially avoid any electromechanical systems which have not already been highly developed by the commercial sphere.

4) Use sophisticated electronic techniques, especially microprocessors, wherever possible so long as the implementation can be accomplished through LSI (Large Scale Integrated circuit) technology. It is increasingly the case that the sophistication of electronic circuits is no longer a significant cost-determinant provided that sufficient market exists to justify the use of "chip" techniques.

5) Aim for a system which is both sufficient (to satisfy the user) and also minimal (does no more than it has to do in order to be cost-effective).

These have certain consequences which we can cite out of context. Audio cassettes can be employed as a storage medium, even for text and graphics, because of CSTV techniques. It provides certain disadvantages, as does the use of the telephone line, such as non-instantaneous accessing of information. However, speed of delivery may still be adequate when related to the very limited capability of human beings to absorb and assimilate information. A "browsing" capability is very important but it is probably not necessary to access a page of text in 1/60 sec. if it takes the person 30 seconds to read it. An ordinary audio cassette, 60 minutes long and costing under $1, can contain about 600 pages of black-and-white text with each page being presented on the screen about 6 seconds after the appropriate frame is located. Half-tone pages would reduce the number of pages and increase the time required to "read in" a page.

System Description We shall make reference to the schematic diagrams in Figures 5,6 and 7, where we have divided up the functioning of our Remote Library Service System into Production & Storage (at the primary and secondary library source locations), Distribution (either via the physical delivery of cassettes or by communication to and from secondary and tertiary library locations) and Display (at a local library location or in the user's office).

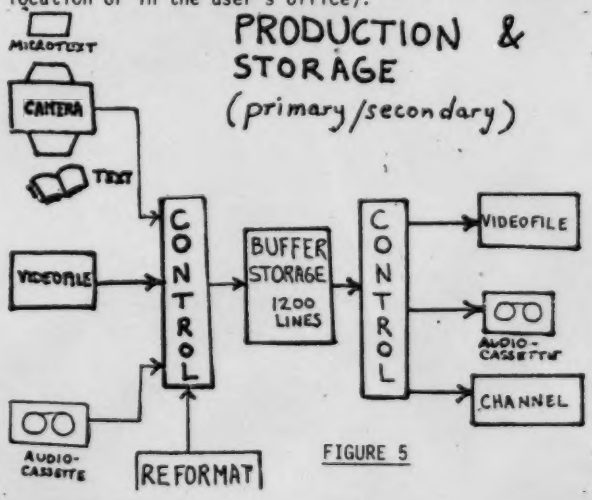


FIGURE 5

Figure 5 shows the elements of the production system. We cannot avoid the intervention of a human to get the material and bring it to a high-resolution camera system ONCE. Input from a non-electronic source can be either from the original text or from a micro-text copy. At the primary library location, the material whould then always (in parallel) go into permanent (but erasable) form onto a computer-addressable videofile which is a dense, high-speed-access system for subsequent accessing of information. Individual pages can be accessed for "professional fair use", providing one means of accommodating to the copyright law. Alternative outputs would be onto audio cassettes or onto a channel for transmission in real time. Alternatively, of course, transmission could take place off-line to utilize idle channel capacity, for example, at night. The "Reformat" control would provide a simple means of accommodating pure-text to the 40-characters-per-line capability matched to 525 line NTSC-TV standards. This would permit straight-forward display on ordinary TV sets.

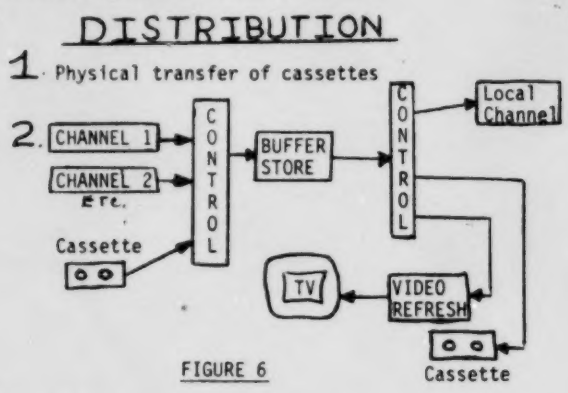


FIGURE 6

Figure 6 refers to the channel matching function of the secondary, etc. distribution centers. The material could be viewed in real-time on a TV display (either high-resolution or ordinary TV), an audio-cassette made or immediately retransmitted over a local channel. Thus, depending on the frequency of usage, a secondary library might find it advantageous to have a broad-band link with a primary library location and retransmit the needed information over local telephone lines to a tertiary library or to an individual user.

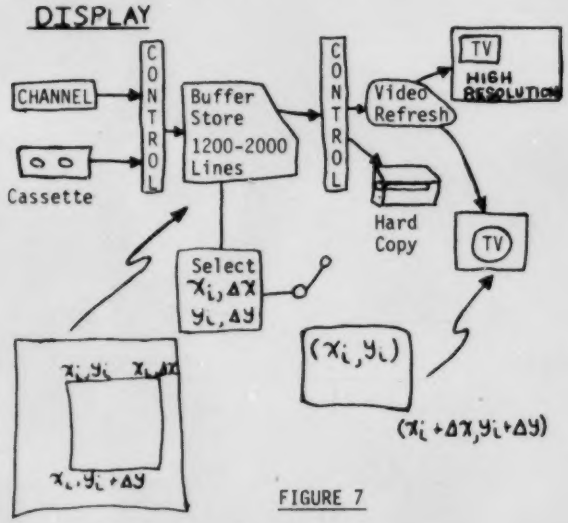


FIGURE 7

Figure 7 refers to the display capability which could exist at any of the possible locations: primary, secondary, etc., or individual user. The display could be on a high-resolution TV monitor (justifiable only at primary or secondary locations) or on an ordinary TV set, which would suit the individual user's office. Alternatively, there could be an off-line provision for making hard-copy for those of us who find new modes difficult to adjust to. The "joy-stick-mode" of operation permits the scanning of high-resolution material on a low-resolution display. The analogy is to a microscope: the TV set is used as a microscope-viewing window for the material stored in the buffer memory.

What is envisaged here is a rather dramatic departure from present modes of operation. It is believed that the individual user may be able to adapt to a mode wherein the documents are maintained in "softcopy format" on audio-cassettes. The full spectrum of devices developed for the home entertainment market could be employed here. The display, of course, is envisaged to be via TV monitor. However, pages could be referenced (by number and automatic forward-reverse reel control) and it is even possible to make written notations on the individual's copy of documents using a simple light-pen arrangement.

Certain human-factors and economic studies should be undertaken before these millennial developments can be expected to be available. However, the flexibility of these ideas militates in favor of some system such as this being produced in the not-too-distant future.

FOOTNOTES

* Copyright 1977

1. Ohlman, H.M., "Communications Media and Educational Technology: An Overview and Assessment with Reference to Communications Satellites", Washington University Report No. (R)T-71/1, (232pp) May 1971, (ERIC No. ED 053 540); Johnson, G.A. & R.J. Bushnell, "Some Behavioral Aspects of Interactive Still Frame Television", Proc. IEEE Int. Conf. Communications, Seattle Wash., June 1973 pp 6-9-6-14; McManamon, P., "Technical Implications of Teleconference Service", IEEE Trans. on Comm., Jan 1975 pp 30-38; George, D.A., D.C. Coll, S.A. Patterson, and P.D. Guild, "Video Via the Tele-Phone", The Telephone in Education Conference, Univ of Wisc., Ext, May 27-28, 1976.

2. Travers, R., "Research and Theory Related to Audio-Visual Information Transmission". Office of Education, OES-16-006 (1967)

3. Cohen, W.C. and S.L. Meyer, "Development of the Educational Uses of Slow-Scan Televideo", Biosci. Commun. 1, 169 (1975); Meyer, S.L., "Extending the Reach of the University with Narrow-band Telecommunications: The Present & Potential Uses of Slow-Scan Televideo for Continuing, Off-Campus Education", Paper commissioned by Alfred P. Sloan Foundation, Oct 1975

4. Meyer, S.L. and D. Brown, "A Review of Available Technology for Narrow-Band Transmission of Visual Material", Biosci Commun. 2, 38 (1976)

5. Overhage, C.F.J., "Project Intrex: A Brief Description", Mass. Inst. of Technology, Cambridge, Mass., 1972

6. Paterson, D.G. & M.A. Tinker, "How to Make Type Readable" (New York: Harper and Row, 1940); Burt, Sir Cyril, "A Psychological Study of Typography" (Cambridge; Cambridge University Press, 1959)

AN AUDIO VISUAL DEVICE DESIGNED FOR TELECONFERENCING USE

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The growth of teleconferencing for education has created the need for an adequate and compatible audio-visual delivery system. The evolution in the design of such a product must consider a large number of requirements and it would be contributory to review some of these parameters.

The effectiveness of a teleconferencing approach would be substantially improved with a controlled, randomly accessible visual that could be linked to the audio presentation.

Visual formats, including film strips, slides, motion picture film, video tape and microfiche were examined. Film strips are comparatively difficult to produce, are fragile and are not readily randomly accessed to individual frames. While slides offer excellent visual quality and are an existing broadly used standard, they are comparatively expensive in good quality reproduction, bulky and inconvenient to store and ship.

Other detriments are, the ease with which presentation sequence is lost on remote controlled units and the requirement of 100% manual inspection to insure program integrity at every reissue. Slides are designed primarily for linear presentations and random accessibility is slow and less than positive.

Video tape and motion pictures are comparatively expensive to produce, store, ship and retrieve while also requiring constant inspection to insure the integrity of program content.

Our evaluation concluded that an international standardized microfiche would be the most cost-effective visual media for teleconferencing. Microfiche can readily be made from existing transparencies and original art and print materials. Microfiche offers slide-like color quality with excellent resolution combined with easy handling, storing and mailing. Also, absolute program integrity is always assured with the use of a microfiche.

Requirements For Effective Programs

Effective teleconferencing systems programs call for active student participation and repetition.

Anything to involve the participants in more than just listening is desirable. The stimulus-response theory suggests that the learner should be an active rather than a passive viewer or listener. This involvement could take the form of asking questions or actually assigning physical tasks. Effective programs will take this into consideration and the design of audio visual equipment for teleconferencing must accommodate this requirement.

Repetition, in the heretofore primarily audio medium of teleconferencing, while desireable for students, has been difficult to provide in an interesting format. The need for repetition has therefore also become an equipment requirement.

The REVOX Audio-Visual System

REVOX of Switzerland has taken all the above into consideration in the design of a randomly accessible audio syncronized microfiche projector system that can be controlled by means of discrete signals sent over telephone lines.

The REVOX "AUDIOCARD"-TM System uses a standardized microfiche (known as the Cosati-18 format), carrying individual images in the form of 60 photographic frames arranged in 5 rows of 12 frames each, on a flat piece of film. That film is placed in a freely moveable carriage, with the position of every single image being determined by a coded address signal.

The address signals are used to control the movement of the film carriage in the horizontal directions by means of stepping motors, thereby determining and maintaining the exact position of each frame in the optical axis of the projector.

Compared with the capacity of the slide projector, the microfiche system provides a saving in space of some 98%, while in addition to this, duplicating costs are lowered by a factor which may vary from 10 to 20.

Teleconferencing program requirements may vary the demand for the visual display from direct control by the lecturer to control by pre-recorded audio programs. To satisfy both requirements the projector is so designed that the search for a image on the microfiche can be initiated either by remote manual control or the pre-recorded address signal on the tape.

The Audio-Visual System Components

Before presenting technical data it would be advisable to get some idea as to the general appearance of each component in the system.

The teleconferencing controlled visual presentation can be accomplished either by the "AUDIOCARD" MONITOR or PROJECTOR.

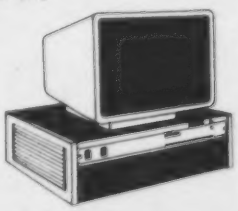


Figure 1

Figure 1 - The"AUDIOCARD" MONITOR is suitable for small groups and can be used in normally lighted rooms.

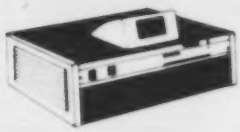


Figure 2

Figure 2 - The "AUDIOCARD" PROJECTOR has a short focal length and is used for front viewing in a normally darkened room.

Figure 3

Figure 3 - The "AUDIOCARD" remote control ENCODER serves two fundamentally different purposes. First of all it makes direct selection of images on either the MONITOR or PROJECTOR and secondly it is used to position control pulses on the tape program when a sound syncronized audio visual presentation is desired

The special feature of the "AUDIOCARD" process is that each coded impulse is assigned to a specific picture on the microfiche.

Figure 4

Figure 4 - The Audio Unit. Any tape or cassette recorder may be used as the sound recording medium, provided it has two independent parallel sound tracks. REVOX has developed a specially designed cassette recorder for this purpose.

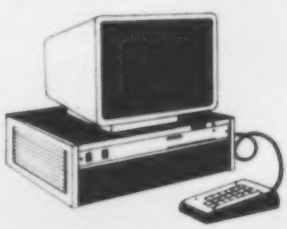


Figure 5

Figure 5 - Shows the MONITOR and ENCODER as they could appear at any location in the teleconferencing network.

Essentially one ENCODER could drive all the MONITORS or PROJECTORS in a teleconferencing network.

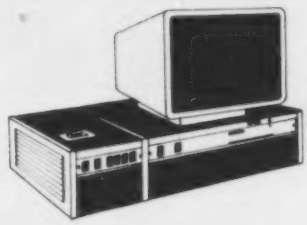


Figure 6

Figure 6 - Indicates a MONITOR with pre-recorded program for a fully automatic operation. This operation can be remotely started or stopped through the network.

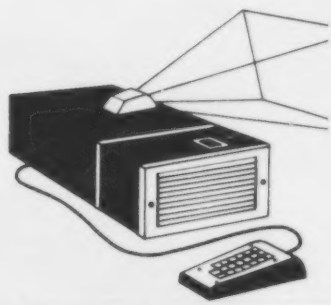


Figure 7

Figure 7 - Shows a fully automatic PROJECTOR presentation with the added facility to take control of a presentation by use of their own ENCODER for immediate recall of any picture.

Microfiche - The Visual Medium

The microfiche as used with the "AUDIOCARD"-TM represents an economical standardized image carrier of postcard size. This visual medium is a standard COSATI, 60 image microfiche with each image measuring 11.75 mm x 16.50mm with an overall size of 105mm x 148mm or approximately 4-1/8" x 5-7/8".

As previously indicated, REVOX selected the microfiche because it represents a complete work unit (presentation) and is the most compact image carrier for visual information.

Other important reasons in favor of microfiche are that the color and conventional continuoustone film material required, can be processed anywhere throughout the world and it lends itself to the original horizontal format used almost exclusively for

visual presentations, such as film strips, movie films, video and approximately 80% of all slide presentations.

The usefulness of the microfiche format is further enhanced by selection of an international standard referred to as A6, COSATI, as determined by the International Organization for Standardization (ISO) publication 2707 dated November 15, 1976.

Microfiche Production

To complete a total audio-visual delivery system REVOX, in close cooperation with the ALPA Camera Company of Switzerland, developed a master photographic package to effectively create microfiche masters from original materials and existing transparancies.

The ALPA-MASTER System, utilizing standard 35mm Ektachrome film, photographs the input material such as slides or paper art work and develops from this roll of film a strip microfiche master ready for duplication by the standard Kodak E-6 color contact print method which is internationally available.

Utilizing the ALPA MASTER, all existing teleconferencing visual materials and programs can be readily converted into microfiche masters that can be reproduced in amounts of 50 or more for as little as $1.50 per fiche. Even greater economies can be achieved as the number of fiche copies increases.

The economies that can accrue in the handling, storing, filing, retrieving, mailing and returning of microfiche as compared to slide trays or other visual materials,become self-evident.

TECHNICAL AND PERFORMANCE SPECIFICATIONS

Technical data on the REVOX "AUDIOCARD", an audio visual delivery system for teleconferencing is as follows:

1. Image Carrying Medium

The image carrying medium is a COSATI color microfiche of the International Standard A6 containing 60 images with an overall size of 105mm by 148mm.

2. Frame Rate

Depending on the program, the images may be called out for projection, singularly (maximum access time is 0.8 seconds) or in sequence for a stroboscopic effect with up to 4 pictures per second. Every kind of change in projection is possible including random individual showing, repetition or ommission.

3. Coding Functions

Besides specific codes for each of the 60 images there are coding possibilities for different auxilary functions such as split screen, fade-in, fade-out, close shutter, tape stop, repeat tape, rewind and stop, and eject microfiche. Computer interface through the ENCODER is also available.

4. Multi-Circuit Switching

Several display units in any combination of REVOX MONITORS or PROJECTORS may be switched with only one coded input.

5. Audio Control of Programs

If desired in a teleconferencing program, any tape or cassette recorder may be used as the audio and visual code medium, provided that it has two independent sound tracks. The code, recorded on the tape, will always select the specific image.

6. REVOX MONITOR

The REVOX MONITOR which is a rear screen color microfiche projector, has the following specifications:

a. Functions Of The MONITOR are controlled by the ENCODER and include lamp on-off, verticle half-frame covering right or left side of screen, eject fiche, re-enter fiche, fade-in, fade-out and complete random selection of any image.

b. The Projection Lamp is a Halogen projection lamp with a two pin socket rated 12 volts at 50 watts. This lamp can be a GE 1400 or Phillips 7025 or Osram 64610 or Mallory PN 120. Because of unique protective controls, lamp life is approximately 150 hours.

c. The Power requirements are 110 volts, 50-60 cycles AC with a consumption of 150 watts and fuse protection of 2 Amp SB.

d. The Screen is daylight, rear projection, reflection-free measuring 8-1/4" x 11-5/8" or 21cm x 29.7 cm.

e. The Dimensions are:

Width	- 17-1/4" or 43.6cm
Depth	- 16-1/4" or 41.0cm
Height	- 16-1/2" or 41.8 cm
Weight	- 33 lbs. or 15kg

7. REVOX PROJECTOR

The REVOX PROJECTOR which is a front screen color microfiche projector, has the following specifications:

a. Function Of The PROJECTOR are controlled by the ENCODER and include lamp on-off, vertical half-frame covering right or left side of screen, eject fiche, re-enter fiche, fade-in, fade-out and complete random selection of any image.

b. The Projection Lamp is a Halogen Projection Lamp with a 2 pin socket rated 12 volts at 100 watts. This lamp can be a GE 1401 or Phillips 2023 or Osram 64625 or Mallory PN 160. Because of unique protective controls, lamp life is approximately 150 hours.

c. The Power requirements are 110 volts, 50-60 cycles AC with a consumption of 200 watts and fuse protection of 2 Amp SB.

d. The Projected Image is a front screen projection that can from a 6 foot distance offer a screen size of 3'6" x 2'4" (or at 2m an image of 1m x 0.75m). Actually the PROJECTOR can offer a clear crisp image up to 5 feet wide if desired.

e. The Dimensions are:

Width - 17-1/4 or 43.6 cm

Depth - 16-1/4 or 41.0 cm

Height - 8-3/8 or 21.2 cm

Weight - 28.5 lbs or 13 kg

8. THE REVOX C-88 CASSETTE RECORDER

The C-88 Cassette Recorder is a versatile remotely controllable 2-track unit. The control logic permits changing from one function to another without having to use the stop key, and with automatic level control, offers the advantage of functioning perfectly with every audio input.

With an interface accessory, the C-88 can be coded to automatically control stop tape, rewind tape and stop, repeat a section of the program, and rewind tape to the beginning and play.

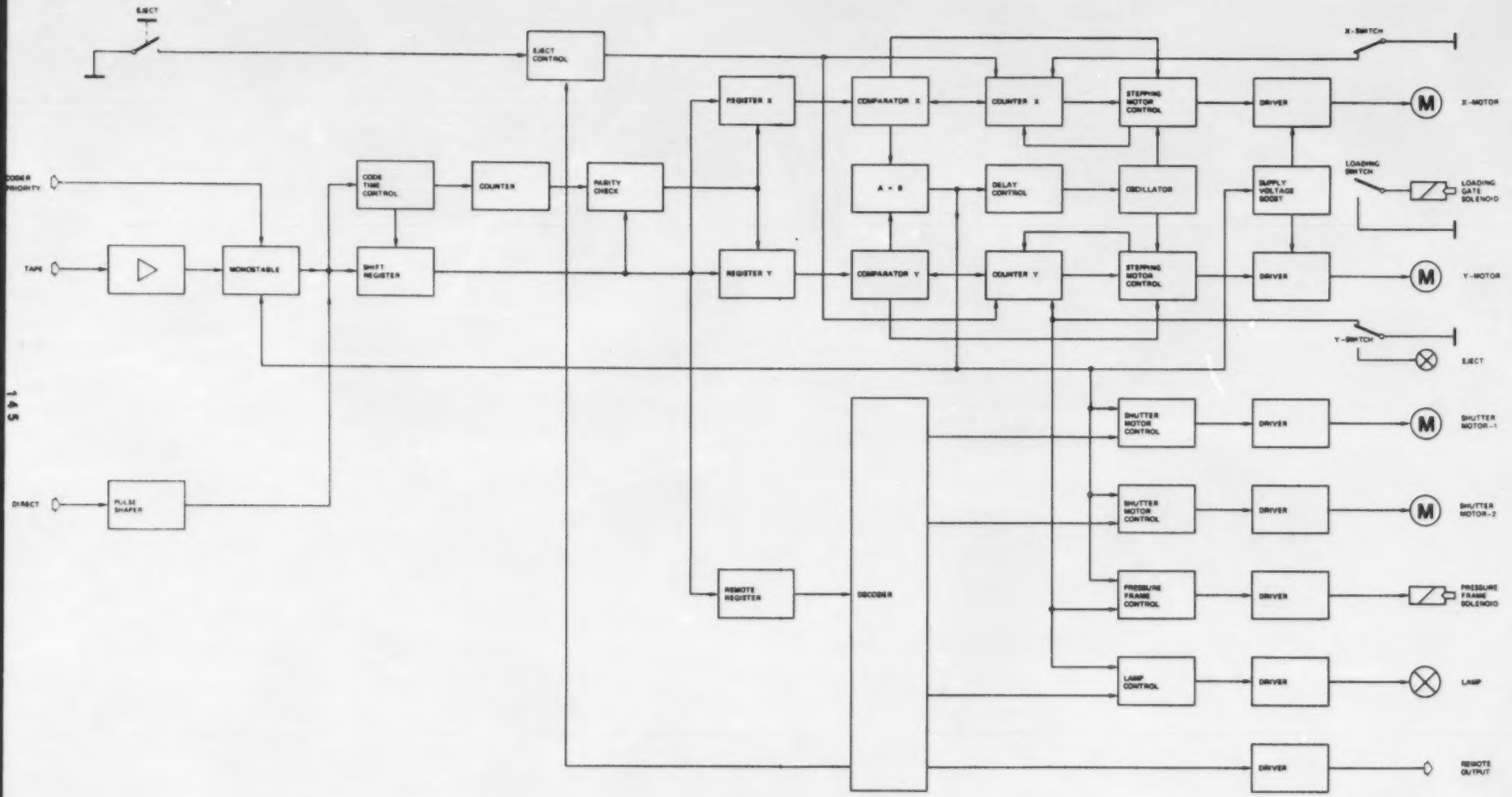
Using the ENCODER and a microphone the C-88 can record a full audio visual program with the audio on one track and the function code simultaneously recorded on the 2nd track. Further specifications are as follows:

Track	- 2/2
Tape Speed	- 1-7/8 ips normal 7-1/2 ips fast copy
Frequency Response	- 100 to 12000 Hz + 2/-3dB
Signal to Noise	- 53 dB, unweighted
Distortion	- 3%/0 dB
Flutter and WOW	- +/- 0.12% at 4.75 cm/s
Fast Forward/Rewind	- for C60 is 60 sec.
End of Tape Sensing	- opto-electronic
Power Requirements	- 110 volts AC,50-60 cycle
Dimensions	- Width 8-1/4" or 21.0 cm
	Depth 16-1/2" or 42.0 cm
	Height 5-5/8" or 14/3 cm
	Weight 16-1/4 lbs or 8kg

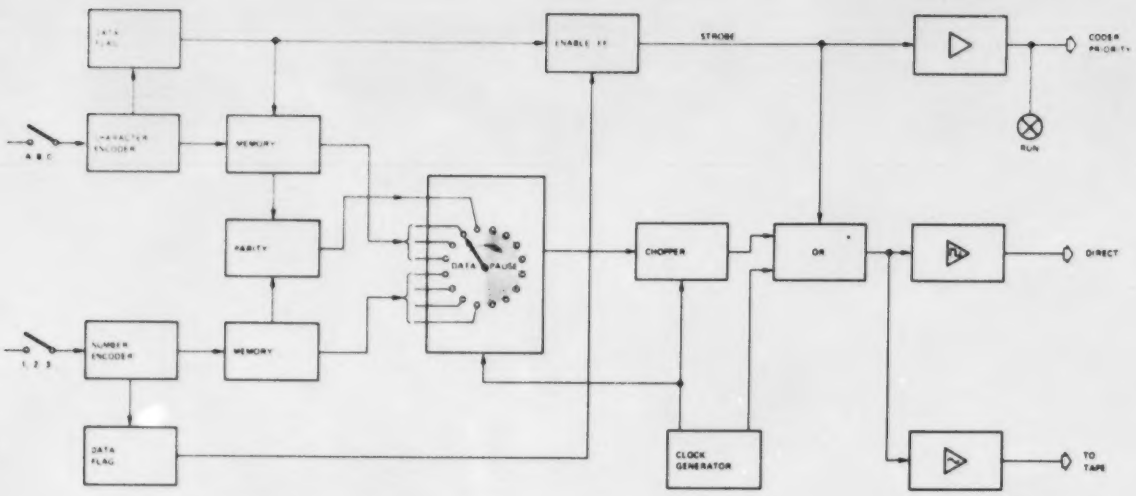
9. REVOX ENCODER

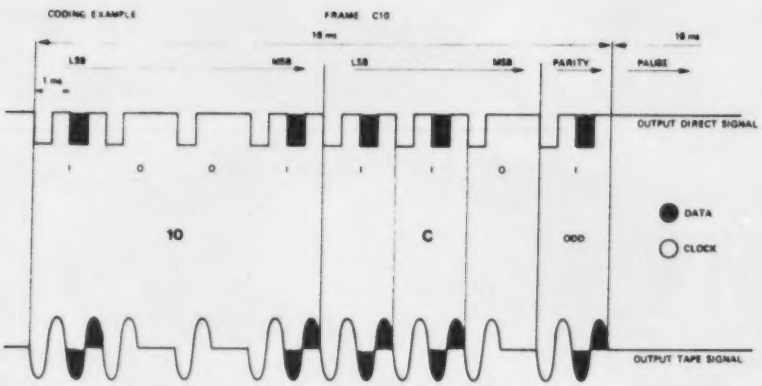
The REVOX ENCODER is designed to emit a continuous binary address code which can drive either the REVOX PROJECTOR or MONITOR and be recorded on any 2 track magnetic tape recorder. This discrete code brings forth a specific image and can further instruct the PROJECTOR or MONITOR in all the functions referred to previously.

It is this continuous binary address code that can be transmitted through teleconferencing and permits total control of the visual presentation aspect of teleconferencing programs.



10. "AUDIOCARD" - MONITOR BLOCK DIAGRAM





BINARY		BIT	BINARY		BIT
Address	BCD		Address	BCD	
1	0	0 0 0 0	R	0	0 0 0
2	1	1 0 0 0	A	1	1 0 0
3	2	0 1 0 0	B	2	0 1 0
4	3	1 1 0 0	C	3	1 1 0
5	4	0 0 1 0	D	4	0 0 1
6	5	1 0 1 0	E	5	1 0 1
7	6	0 1 1 0			
8	7	1 1 1 0			
9	8	0 0 0 1			
10	9	1 0 0 1			
11	10	0 1 0 1			
12	11	1 1 0 1			

11. "AUDIOCARD" - ENCODER BLOCK DIAGRAM

AUTOMATIC TELEPHONE ACCESS LIBRARY STORAGE SYSTEM

Rodney M. Sabick
Project Director
International Tapetronics Corporation

During the past few years, library storage via automatic control has received a great deal of attention. Previously, telephone access of library storage information has been limited, primarily, to operator-type intervention. The Telephone Access Library Storage (TALS) System performs the following sequential tasks:

1. Automatically answers the telephone calls.
2. Plays a response message to the caller.
3. Decodes either rotary dial or Touch-Tone[1] entries made by the caller for the desired library message.
4. Retrieves the audio tape cartridge and plays the requested library message.
5. Disconnects the telephone line upon completion of the requested message.
6. Provides statistical information concerning system usage.
7. Returns the audio tape cartridge to its proper storage location.

The standard system consists of the following:

1. Accommodation of 1024 audio tape cartridges (each with up to 10 minutes of message information at 7 1/2 IPS tape speed; 20 minutes at 3 3/4 IPS).
2. Control of up to 12 telephone lines.
3. Loading, unloading, replacing and processing audio tape cartridges via an electronic handheld keyboard.

In addition, the TALS unit can be expanded to twice its basic capabilities.

Operational Description

The controller answers an incoming call by connecting a response message to the telephone line through the telephone interface network as shown in figure 1. This message informs the caller that he has gained access to the TALS system and instructs him to enter the digits corresponding to the desired library message. At the completion of the message, the controller disconnects the response message device and connects the telephone digit decoder to the line.

The telephone decoder converts rotary dial pulses or Touch-Tone[1] frequencies into digital representations. The controller accepts this digital information and determines the availability of the requested message. A time limit is set to enter the four digit message number. Fifteen seconds are alloted until the first digit has been entered and seven and one-half seconds for the three consecutive digits. If the entries are not made within the time constraints, a busy signal occurs for two seconds prior to disconnection of the line.

If the incoming request is an available library cartridge, the controller directs the appropriate mechanical handling equipment to load the requested cartridge into an available tape cartridge machine. This loading process requires approximately five seconds. If the message is not available, either because the requested audio tape cartridge is currently in use or not within the system, the appropriate explanation message cartridge is loaded. The controller then starts the appropriate cartridge tape machine and connects the requesting telephone line to its audio output via the telephone interface network.

When the message ends, the controller disconnects the tape cartridge machine from the telephone line and sends out the appropriate instructions to return the audio tape cartridge to its proper storage location. The controller also provides logging information including the message number, time of day, date, telephone line number, cartridge machine number, misdials, whether it played, whether it was busy, whether the caller listened to the complete message, and when total system usage occurs.

Operational Features

The controller network is the heart of the TALS. It consists of a microprocessor and the associated interfacing elements which enable it to control the complete system. The processor handles the mechanical aspects of the system while simultaneously keeping track of all the incoming calls, tape cartridge machine status, and the input-output load keyboard.

The tape machines used are broadcast cartridge type that meet the specifications of the industry. These cartridge machines use endless loop tape cartridges that require no rewinding. The cartridge machines are started by the controller and are stopped automatically by a stop tone on the tape itself.

The telephone Interface network detects the initialization of incoming calls on any telephone line and provides the audio connections between the telephone lines and the audio from the tape machines. Isolation between the telephone equipment and TALS, as is required by the telephone companies, is achieved with optical couplers. The Decoder network consists of a Dual Tone Multiple Frequency (DTMF) decoder and rotary dial pulse decoder. Electronic filters and circuitry decode the DTMF signals into a binary representation.

Rotary dial detection is complex because rotary dial signals are no longer pulses at the receiving end. The central office does not pass the DC pulses generated by the phone, thus making the signal at the receiving end an audio signal created by the make and break of mechanical contacts. The incoming signal appears as an exponential decay and is even more difficult to detect due to inherent central office noise.

Decoding these dial pulses may not be possible through all central offices because of the telephone system that may be employed. Once the station-to-

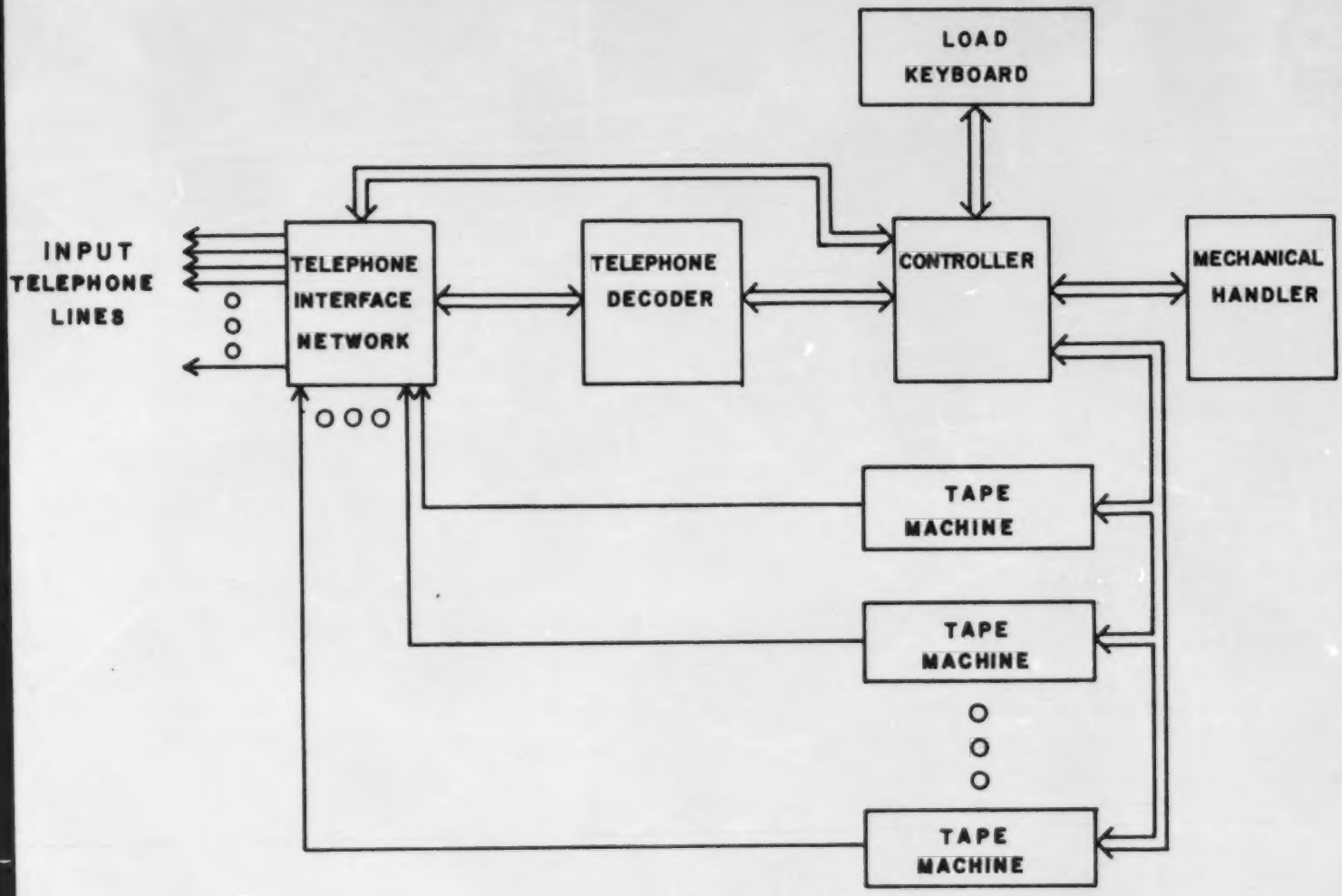


FIGURE I.
SIMPLIFIED BLOCK DIAGRAM OF TALS

station connection is made and the TALS system is accessed, any further dialing of the user's phone may result in the disconnection of the line by the central office. Some central offices will detect post dial pulses as a break in the loop current just like a break in the loop current occurs as a phone is hung up. Some central office systems will work and others will not because of differing loop break detection responses.

Because of the large number of different systems in use, there must be some provision made for people who are unable to access the system with rotary dial sets. Since there is no reasonable expectation of changing the existing phone systems, the caller's phone can be modified or an attendant can be provided for callers connected to non-compatible dial systems.

A DTMF generator can be added to existing dial telephones enabling a caller to enter a message number with a Touch-Tone[1] keyboard. A simple DTMF circuit consists of a 2-of-8 Touch-Tone[1] integrated circuit and a few other inexpensive components. However, the practicality and expense of either an attendant or dial telephone modifications needs further research.

The Mechanical Handler includes two storage drums and a Lift-Turn-Retrieve (LTR) mechanism, as shown in figures 2 and 3. The storage drums are arranged in 16 columns of 32 rows each for a total of 512 cartridge storage locations per drum. Each drum is capable of rotating in either direction to any one of the 16 column positions. The LTR unit is an elevator type mechanism capable of lifting, rotating and retrieving any audio cartridge. The LTR positions to any one of 32 vertical locations, rotates to either the drums, cartridge machines or the load chute, and moves a cartridge to or from a cartridge tape machine or storage location. The controller activates the motors that move these mechanisms to the desired positions facilitating the movement of a cartridge from one location to another.

A load chute and unload basket are provided to facilitate the addition of cartridges into the system or removal of old cartridges. This loading-unloading process is done in conjunction with a small keyboard and associated LED display. To load cartridges into the system, the cartridge is placed in the load chute and its desired storage location is entered into the keyboard followed by keying the LOAD button. Up to seven subsequent entries may be made without waiting for the cartridges to be physically loaded. If a load instruction is given to load a cartridge to a storage location which has already been assigned, the LED display will indicate the error. If another location is desired, the CLEAR button is keyed, followed by keying in the new location and the LOAD button. If it is desired that the original requested location be used, the REPLACE button is keyed. This activates the unloading and replacement of requested location with the desired audio cartridge. Cartridges may be unloaded by entering the cartridge number followed by keying the UNLOAD button. To accommodate popular cartridges, up to five cartridges with the same message can be equated. These cartridges are equated to each other by entering the cartridge number and keying the EQUATE button. This process is repeated for the cartridge and/or cartridges that are to be equated.

An artist's rendition of the TALS system is shown in figure 4. The Systems are presently being built and used in the highly competitive radio broadcasting industry.

Summary

The TALS system, as described above, does have the potential to fill the need for an automated library storage system. The major obstacle, (callers with a dial phone connected to a non-compatible telephone system) needs additional examination. However, as the System is put to work in the Broadcast Industry, additional technical innovations and refinements will provide an answer to this difficulty.

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[1]Touch-Tone is a registered trademark of AT&T.

COMPUTER CONTROL OF TELECONFERENCING

James R. Potter
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Indiana Higher Education Telecommunication System

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Background

The Indiana Higher Education Telecommunication System (IHETS) was established in 1967 by action of the Indiana General Assembly to provide a structure through which the institutions of higher education in the state, public and private, might work together to satisfy their joint needs for telecommunication services. During the ten years of IHETS' existence several telecommunication systems and services have been established and operated to the benefit of its member institutions. These systems and services include:

- Closed-circuit, switched television network*
- Dial tandem telephone network
- Radio Network
- Teleprocessing* (automatic tape delay television system)
- Teleresponse* (student talkback system)
- Shared Telpak purchase agreement (voice and data circuits)
- Hard-copy message network*

IHETS is currently setting up a pilot voice conferencing network which will build on the Wisconsin experience with ETN and SEEN. We are also beginning a two-year program to develop and evaluate a tape television system called Tele-Library.

Network Design Philosophy

IHETS component networks and technical systems have been designed and developed around the following basic principles:

Reliability

Educational support systems must be reliable. They must not only work the first time a professor attempts to use them, but they must also work day after day, becoming as dependable as the heat and light systems in a typical classroom. Faculty confidence in media is in short supply, and there is usually good reason for this lack of confidence. New systems must earn faculty acceptance.

Flexibility

System flexibility is a top priority. In designing systems we attempt to anticipate future possibilities in addition to satisfying the immediate need. Systems are designed to keep as many options open as possible for future changes and growth.

Time Sharing

Sharing the use of a facility is a valuable technique for reducing the unit cost of system operation. Where user needs can be blended to share time on a facility, the efficiency of that facility can be enhanced. Hardware systems can also be integrated in some instances to minimize costs.

Users Need Help

The most flexible, reliable system will fail if users are not provided with the information and support necessary to assure proper system use. Users must be able to communicate with someone who will help solve technical and operational problems when these occur. The "call for help" procedure must be clear to all users and system personnel must be available to follow through on all problem situations immediately.

System Operation Continuously Monitored

System personnel must play an active role in monitoring system performance. Complete reliance on user trouble reports is not sufficient to assure network reliability primarily because user reports are most likely to occur immediately preceding or during intended system use, leaving little or no time to correct the problem before programming is affected. Moreover, users are often reluctant to report minor dissatisfactions with system performance. These then frequently become continuous irritants to users, fostering a negative attitude toward the technology. An automatic monitor and alarm system, however, will signal faults instantly, increasing the likelihood of correction with no loss of programming.

Routine Network Functions Automatic When Feasible

By automating routine switching and monitoring functions, technical personnel are left free to work with users when problem situations arise. Machines can usually perform routine operations better than people; on the other hand, a person experiencing difficulties with a technical system usually prefers to have a person come to his aid instead of a machine.

Microprocessor Control of Teleprocessing and Teleresponse

The technical details of two network sub-systems, Teleprocessing and Teleresponse, are described below. These systems were developed by the IHETS staff using microprocessor-based automatic control systems; an integrated approach has been taken to their design. They share intercity telephone lines and some electronics as well as the microprocessor control system. Plans for the future envision additional tasks for the control system, such as the control and monitoring of several supplemental television distribution facilities (ITFS and CATV) being constructed to extend the television network to additional service locations.

Microprocessor Use with Audio Conferencing

The ETN-type network currently starting as a pilot project will begin as a copy of the Wisconsin system. Later phases of this project, however, will involve the

*Systems marked with an asterisk (*) in the above listing are controlled by computer.

use of microprocessors to add such features to the network as:

- Automatic monitoring of system operation
- Conference confidentiality
- Student feedback systems (tests, attitudes, etc.)
- Electronic graphics
- Hard-copy messages and automatic data collection
- Automatic network segmentation

Just how many of these features will be of value remains to be determined as the network develops. Until then we are keeping options open for adding these and perhaps other features as technology matures and needs become apparent.

Teleprocessing

The Teleprocessing project was initiated by IHETS to develop a means of increasing television network usage and to reduce scheduling conflicts arising from the limited number of network channels available, especially during daily peak usage periods.

Teleprocessing is a system of remotely-controlled video record/playback stations (Teleprocessors) which function under the control of a centrally-located computer. Since a substantial portion of the programming carried on the network is not live, Teleprocessing allows us to pre-transmit some material when the net work channel is available, record it on a Teleprocessor at the intended receive location(s), then play it back at the regularly scheduled time when the network channel is being used for another, possibly live program. The system is operated entirely by remote control and appears, in effect, as a second incoming network channel at the receive location. FIGURE 1 illustrates a typical application of Teleprocessing to alleviate a scheduling conflict.

System Design Objectives

We determined that each Teleprocessor should have the capability of recording and playing back programs of up to two hours in length as well as storing two separate programs at each location. Another goal was full automation of the central control facility so that the IHETS control room operator could enter the Teleprocessing schedule in advance, with actual machine control handled by a computer, in a manner analogous to the operation of the video network itself. Finally, we felt it essential to provide a means of transmitting tally information from each Teleprocessor back to the central control computer to verify proper system operation.

The U-matic 3/4-inch cassette format was ultimately selected for use in Teleprocessing. Use of a one-inch format VTR was considered but rejected because of the increased complexity involved in controlling the rewind and recueing of an open-reel machine. The cassette format was ideal in being a closed system, and the fact that the cassette machines were designed with a minimum of external adjustments, to facilitate their use by non-technical personnel, makes them well suited for operation in a long-term, hands-off environment.

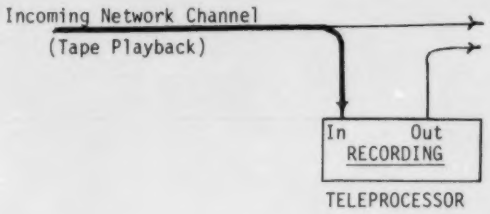
Two VTRs are incorporated into each Teleprocessor to meet our objectives of two-hour record/playback capacity and ability to store two separate programs. Thus, the system as it is presently configured can store two separate programs one hour or less in length or one program of up to two hours' duration. In addition, the use of two VTRs provides a substantial back-up capability in that the system can still function, in a limited mode, if one VTR becomes inoperable.

The VTRs and associated control and signal

FIGURE 1

EXAMPLE OF TELEPROCESSING APPLICATION

Assume that a particular receive location, with only one incoming network channel, desires to receive two separate programs at the same time. If one of these programs is a videotape playback rather than a live broadcast, it can be transmitted at an earlier time when the network channel is not in use and recorded on the Teleprocessor at the receive location:



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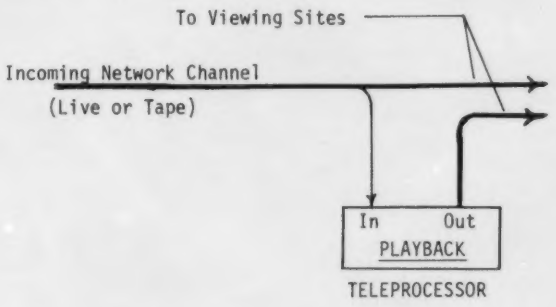
At the regularly scheduled program time, the second program, which may be live or taped, is sent via the regular network channel. Simultaneously, the Teleprocessor is commanded to play back the recording of the first program:



electronics are housed in a locked equipment rack to minimize the possibility of tampering. FIGURE 2 is a block diagram of the Teleprocessor.

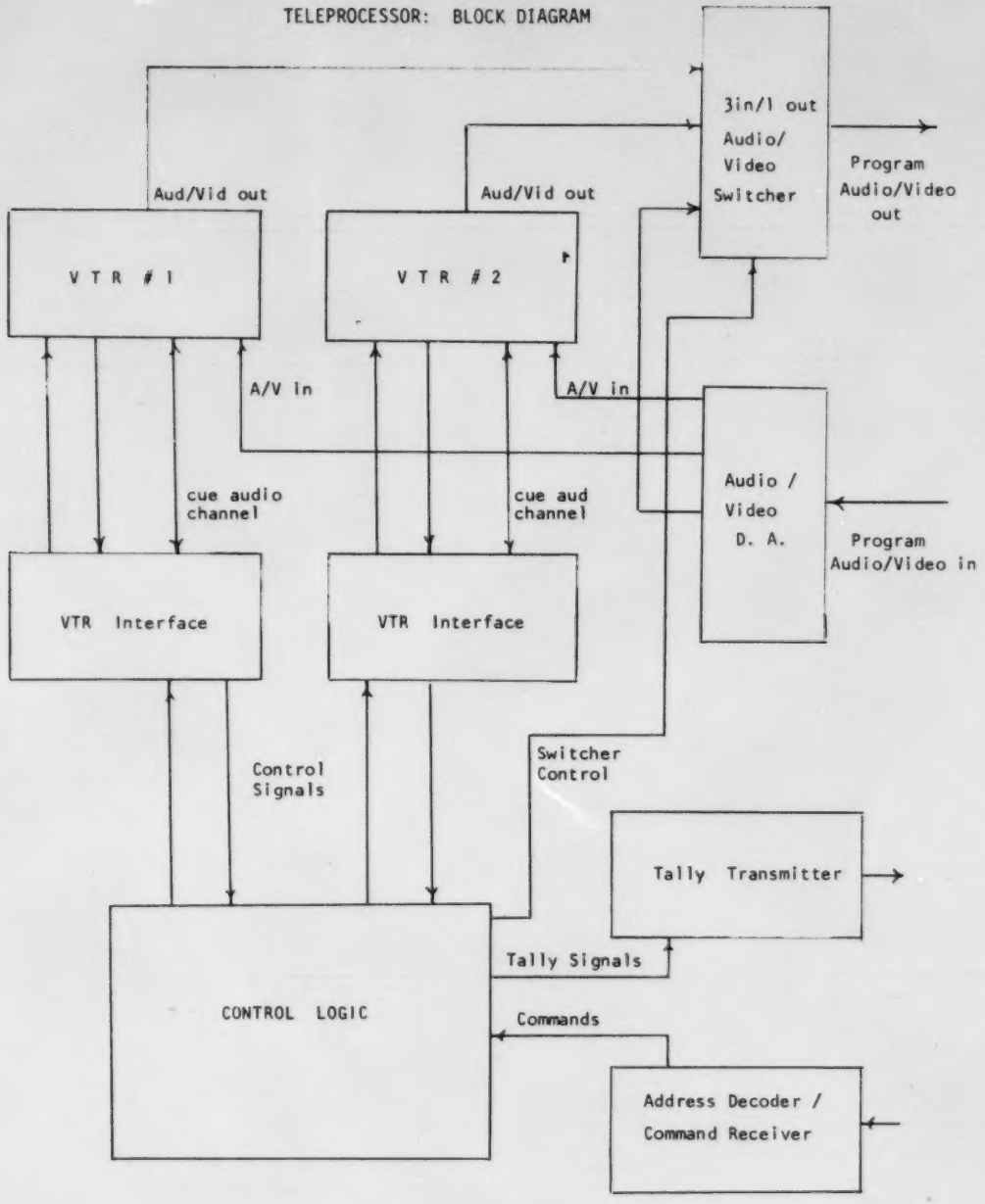
System Description

The program audio and video distribution and switching are handled by the two modules on the right-hand side of the diagram in FIGURE 2. The distribution amplifiers have three outputs: one for each VTR and one which drives the output switcher directly to provide an "E-E" signal during a recording or when neither machine is in use. The two remaining switcher inputs are connected directly to the two VTRs.

Each VTR is connected to the control system via a VTR interface module. The purpose of this module is effectively to mask the unique machine control and status signal characteristics of the particular VTR in order to present a standard signal configuration to the controller. This modular approach, in addition to being a servicing aid, increases the flexibility of the system and allows easy expansion or modification. As an example, the first Teleprocessors used the Sony VO-1600 VTR, while the latest units use the newer VO-2600. All that was required to accomplish this change was the

FIGURE 2

TELEPROCESSOR: BLOCK DIAGRAM



design of a new VTR interface module; the rest of the system remained unchanged. With an appropriate interface module, almost any type of VTR could be employed in a Teleprocessor.

The command receiver/address decoder module receives command and address signals from a dedicated telephone circuit from IHETS. These signals are translated into DC control pulses for the control logic module. The address decoding capability allows any number of Teleprocessors to share a multi-drop or party-line control circuit without interference, which can substantially reduce the cost of the control circuits on large systems. The control format employed by IHETS consists of a series of three audio tones, each tone being one of eight distinct frequencies. The first two tones are for addressing; thus, we can uniquely address up to 64 Teleprocessors with this format. The third tone indicates the type of command.

Again, the modular format allows a relatively painless transition to a new type of control by simply replacing the command receiver module. An appropriate module could control a Teleprocessor via connection to a modem and data link, a separate hard-wired controller, direct connection to a microprocessor, or a simple set of push-buttons for direct local control.

The tally transmitter module interfaces with the transmit side of the control link and translates the

status signals from the control logic module into a form suitable for transmission over the control circuit. We are presently employing a system which uses Touch-Tone[tm] signalling to transmit the tally information. The tally transmitter module may be wired to transmit continuously or to transmit in a "polled" mode whereby the tones are only transmitted upon receipt of a polling command from the central controller. This latter configuration is the counterpart of the address decoder in the receive section in allowing several Teleprocessors to share a common tally circuit.

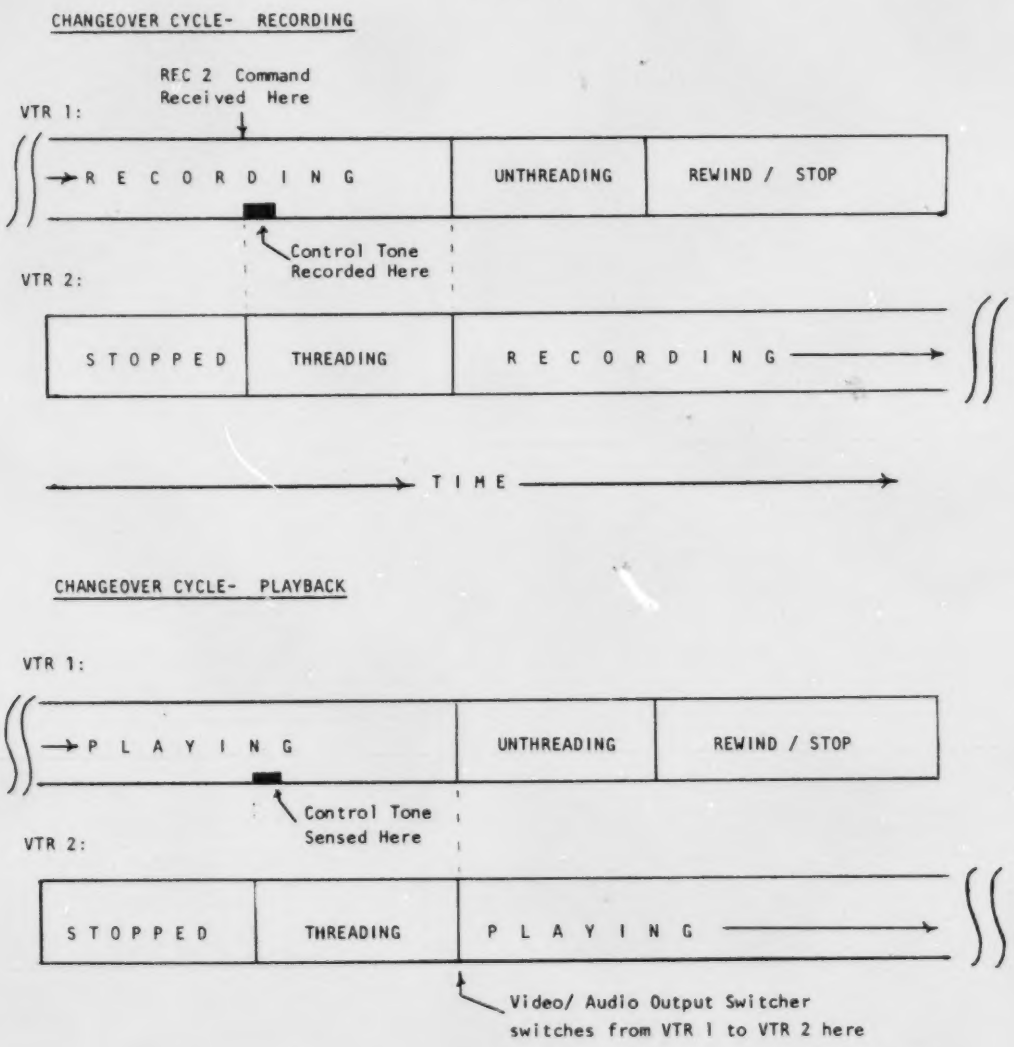
The control logic module links together all system components and contains the logic necessary to supervise the operation of the VTRs in the selected mode. Circuitry in this module generates and detects special control tones on the cue audio track, controls the audio/video switcher, and translates machine status signals into system tally signals for the tally transmitter. In addition, the control logic module contains the logic for controlling the transition between VTRs when recording or playing back a program which is longer than one hour.

Command Summary

There are five machine control commands to which the Teleprocessors will respond. The PLAY 1 command causes VTR 1 to enter a play mode; then, if VTR 2 is running, it is stopped and rewound. The PLAY 2 command operates in an identical fashion for VTR 2. The play commands are only functional if the indicated VTR is stopped prior to issuance of the command.

The RECORD 1 command starts VTR 1 in record mode; then, if VTR 2 is running, it is stopped and rewound. The RECORD 2 command operates in a slightly different manner in order to allow a continuous recording of more than one hour. VTR 2 is started in record mode; at the same time, if VTR 1 is in record, a special control tone is recorded on the cue audio track of VTR 1; then it is stopped and rewound. This control tone, when detected on a subsequent playback of VTR 1, will start VTR 2 in play mode just as if a PLAY 2 command had been received. In this manner, the changeover process which occurred during the recording is duplicated at the same point during the playback so that the viewer sees no discon-

FIGURE 3


CHANGEOVER CYCLE- RECORDING
REC 2 Command Received Here
VTR 1:
R E C O R D I N G
UNTHREADING
REWIND / STOP
Control Tone Recorded Here
VTR 2:
S T O P P E D
THREADING
R E C O R D I N G
T I M E
CHANGEOVER CYCLE- PLAYBACK
VTR 1:
P L A Y I N G
UNTHREADING
REWIND / STOP
Control Tone Sensed Here
VTR 2:
S T O P P E D
THREADING
P L A Y I N G
Video/ Audio Output Switcher switches from VTR 1 to VTR 2 here

tinuity in the program during the transition between VTRs. Since the VTRs are not sync-locked to each other, a brief roll may occur in the video at this point, although this has not been found to be objectionable in practice. FIGURE 3 is a timing diagram that illustrates the series of events which occurs during the changeover process.

The final machine control command is STOP, which causes whichever machine is running to stop and rewind. Since only one VTR may be in operation at any given time, only one STOP command is required. In the case of a STOP command received during a recording, the VTR shutdown is delayed momentarily while a special control tone is recorded on the cue audio track. On the subsequent playback the detection of this tone serves as a signal to the logic to stop the machine at that point so as not to continue playing past the end of valid program material. Note that the STOP command causes the VTR to unthread, then forces a rewind to the head of the tape. Only when it has reached the beginning of the cassette will the machine actually stop. This means that all programs begin at the start of the tape and that any time a machine is stopped we are assured that the tape is fully rewound. Consequently, there is no need to perform a fast-forward operation at any time.

Although the capability of starting a recording at any point on the tape would provide additional flexibility, multiple recordings on a single tape for example, the lack of a practical scheme for "searching" out and cueing a particular program makes this approach impractical. We feel that the design which we have chosen provides a good compromise between hardware simplicity and operating flexibility and possesses sufficient intelligence to be operated by manual or automatic control.

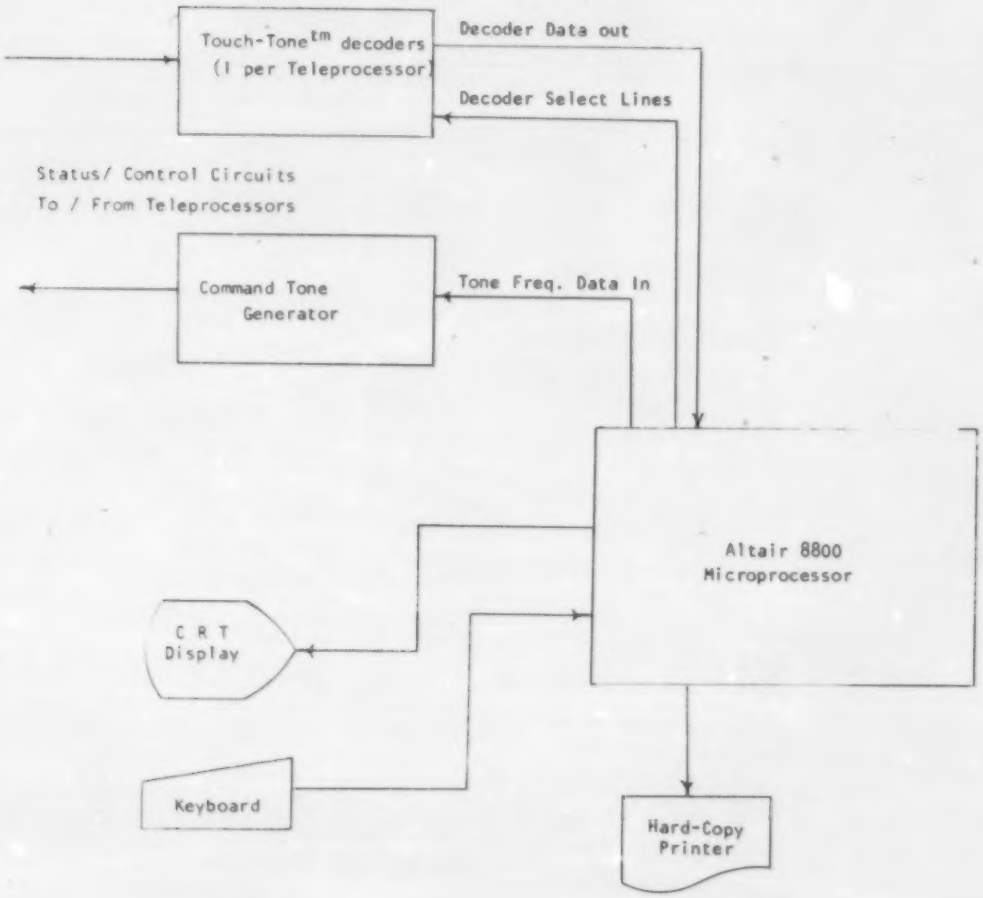
Central Control Facility

Control of all Teleprocessors in the system is handled by a microcomputer located at IHETS in Indianapolis. In addition, a manual control and status panel is kept on-line to serve as a back-up in the event of a malfunction of the primary controller. FIGURE 4 is a block diagram of the control system.

The computer interface to the Teleprocessing

FIGURE 4

CENTRAL CONTROL SYSTEM: BLOCK DIAGRAM



control circuits by means of a bank of Touch-Tone[tm] decoders on the incoming side and by a command tone generator on the outgoing side. In our particular installation we are using a separate tone decoder for each Teleprocessor; this is necessary because certain elements of the control system are shared with the Telereponse (talkback) system. Without this complication it would be possible to handle all Teleprocessors with a single tone decoder, using the polled tally scheme discussed earlier.

Other control system components include a CRT display system and keyboard to facilitate operator interaction and a hard-copy printer which is kept on-line to provide a permanent record of system operation.

The operator enters Teleprocessing requests via the keyboard. Each entry consists of Date, Time, Function (PLAY 1, REC 2, STOP, etc.), and a list of the location(s) where the function is to be performed. The computer maintains a queue of these entries, in chronological order by execution time, and the operator may request a display of the current contents of the command queue at any time.

A real-time clock attached to the microprocessor enables the time-keeping routine to calculate the current date and time of day. This information is always displayed on the CRT; in addition, the control program continuously compares the current time/date information with the time/date information fields in the next queue entry. When this information matches, the computer executes a program which causes the command generator to transmit the proper series of address and command tones to the desired location(s). Simultaneously, the program monitors the incoming tally signals to verify that the Teleprocessor has responded properly. An audible alarm signal alerts the operator if a Teleprocessor should fail to respond properly to a command.

FIGURE 5 represents a typical display of the command queue, with comments to identify the various fields. In addition to this display, the operator may request a display of the current tally signals from all Teleprocessors as well as the status of the tally circuit itself. If no tone, or an illegal tone combination, is being received, the display will alert the operator to that condition.

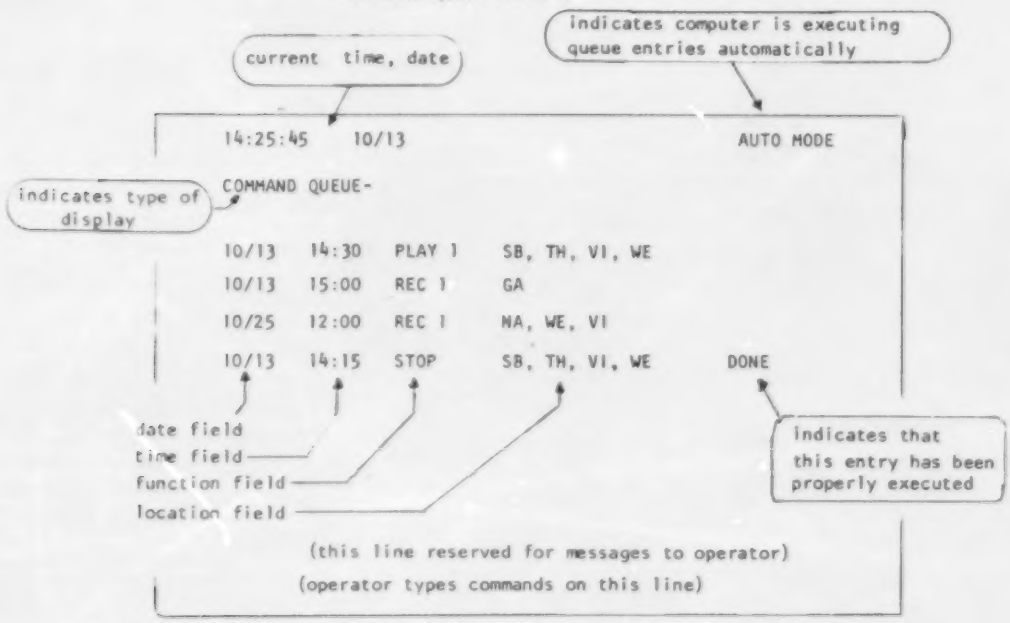
Additional keyboard commands allow the operator to delete or modify queue entries, enable or disable automatic control, reset system time or date, manually control any Teleprocessor, and enter a self-test routine whereby the microprocessor verifies the integrity of the control program operation.

System Software

The main system control program is written in BASIC. A high-level language was selected to reduce the time required for the programming effort. The prices paid for this time-saving are increased memory requirements and a corresponding increase in program execution time. The latter consideration was not significant for the majority of the control functions, but certain time-critical routines (such as time-keeping, tone generation timing, and decoder scanning) had to be coded in assembly language for maximum execution speed.

FIGURE 5

COMMAND QUEUE DISPLAY


indicates computer is executing queue entries automatically
current time, date
14:25:45 10/13
AUTO MODE
indicates type of display
COMMAND QUEUE-
10/13 14:30 PLAY 1 SB, TH, VI, WE
10/13 15:00 REC 1 GA
10/25 12:00 REC 1 NA, WE, VI
10/13 14:15 STOP SB, TH, VI, WE
DONE
date field
time field
function field
location field
indicates that this entry has been properly executed
(this line reserved for messages to operator)
(operator types commands on this line)

The computer sorts the queue entries chronologically and displays them in order of execution time, with the exception of entries which have already been executed- these are shifted to the bottom of the list.

The BASIC software is linked to the assembly-language subroutines and handles such tasks as generating, sorting, and maintaining the command queue; generating and updating the displays; and interpreting and processing keyboard input.

The combination of high-level language and assembly language programming has produced a software system which is quite flexible and easily modified, yet which meets all speed requirements of the system. The Teleprocessing control software, in fact, requires only about half of the available CPU time for execution; thus, the system has the capability of running other tasks concurrently. At the present time, for example, the same microprocessor controls the Teleresponse (talkback) system and an intercity hard-copy message network. A side benefit of having all of these functions performed by a single machine is that the same operator interaction system (CRT and keyboard) can be used to monitor and control all systems, thus minimizing the duplication of hardware that would occur if separate control systems were used.

In addition, the microprocessor hardware, existing peripheral devices, and BASIC software provide a backbone system which allows us to implement new microprocessor-controlled tasks at very little cost.

The Future of Teleprocessing

Our success with the Teleprocessing system to date has convinced us that the approach is a sound one. IHETS is currently applying for a grant from the National Science Foundation for the design and construction of the next generation of Teleprocessing, which we call Tele-Library. The Tele-Library concept envisions a system of multiple cassette record/playback decks--perhaps as many as six--with full random access via a mechanical tape-loader to a large library of videocassettes. Users at remote sites could call out a display of the programs currently available at a particular Tele-Library unit, schedule tape playbacks, enter requests for new programs to be recorded, etc. An internal microprocessor would maintain a table of scheduled playbacks, check for scheduling conflicts, update the library display as new programs are acquired or old ones deleted, and communicate with a central control computer to coordinate the recording of new programs and to forward program requests and messages.

Summary

IHETS presently has Teleprocessors installed at twelve locations throughout Indiana, and in the three years of their existence these Teleprocessors have served to store several hundred hours of program material. We feel that our original design goals have been achieved and that Teleprocessing is a viable alternative to the addition of new network channels.

Teleresponse

Teleresponse is a computer-controlled, one-way, voice talkback system used in conjunction with live instructional programming on the IHETS Video Network; it allows instantaneous direct voice communication between viewers at a receive location and personnel at the originating studio. Teleresponse was designed and implemented by the IHETS staff as a replacement for an earlier system which employed electro-mechanical dialers to communicate over the standard switched telephone network. The old system had several disadvantages: audio quality was often poor, response time was long (typically 15 seconds from the time the push-to-talk switch was depressed until the connection was completed), the equipment was quite bulky and not easily portable, and acoustical feedback often occurred when the talkback unit handset was near the program loudspeaker.

Design Objectives

During the design phase of the Teleresponse project, we placed a great deal of emphasis on simplicity of operation for the classroom user. We feel that talkback can be an extremely valuable tool in restoring much of the rapport between student and instructor which is easily lost when television is employed as an instructional tool. Technical problems such as poor audio quality, complicated operation, and long delays will quickly discourage use of the equipment. Consequently, these areas were studied closely.

System Description

To maximize audio quality and minimize switching delays, Teleresponse employs dedicated, full-period, intercity telephone circuits for the transmission of voice and signalling information. Three main components make up the Teleresponse system: the portable classroom units which initiate the conversations, the master units at each of the campuses which control the classroom units, and the central controller which switches the incoming Teleresponse audio to the desired studio. These components are organized in a "hub and spoke" arrangement as shown in FIGURE 6.

Classroom Units

The portable classroom unit, diagrammed in FIGURE 7, serves as the students' link to the Teleresponse system. Although the system is limited to one-way voice transmission, we employ standard push-to-talk telephone handsets instead of a separate microphone. The telephone handset is familiar to everyone, whereas people are often intimidated by the presence of a microphone. In addition, a telephone handset is held in a consistent manner by each user, while a microphone produces inconsistent user techniques and, consequently, varying audio levels. Each classroom unit is equipped with two handsets on long cords so that they may be positioned anywhere in the room.

The classroom units receive their power over the control pair from the campus master unit; thus, there is no need for a separate power connection in the classroom. Installation is quick and easy, requiring only a single plug-in connection. A rotary switch on top of the classroom unit enables the user to select the studio with which he will communicate. This switch controls the frequency of an internal Touch-Tone™ generator. The generator's output is briefly switched across the line at the beginning of the transmission. Circuitry in the unit controls the timing and switching of the tone signal and handset audio, which are coupled to the line by means of a standard telephone matching network.

The classroom unit also generates a muting signal whenever a Teleresponse conversation is in progress. This signal is used by external muting circuitry in the classroom to reduce the level of program audio driving the classroom speaker by about 12 db to eliminate the possibility of acoustical feedback. Since the system is intended to be used as a push-to-talk/release-to-listen configuration, this momentary reduction in level does not interfere with normal conversation.

Campus Master Unit

FIGURE 8 is a block diagram of a campus master unit, which serves to control and monitor its associated group of classroom units. The master unit converts the DC-signalling indicating on/off status from

the classroom units into tone-signalling suitable for transmission over the telephone line to the central controller. This is achieved by a tone generator which generates a special tone combination (not available as a studio select tone) to indicate to the central controller that the system is not in use. Since this tone combination is continuously present on the line when Teleresponse is idle, it serves as a constant verification that the circuit is functioning.

The master unit provides power to all associated classroom units. When a classroom unit is activated by depressing the push-to-talk button, the DC current flow is sensed by the master unit, and that classroom unit is coupled to the telephone line. At this point, all other classroom units are locked out until the first unit releases the push-to-talk switch, which in turn causes the master unit to switch back to the off-tone generator, thus signalling the end of the conversation to the central controller.

Individual adjustments are provided on the master unit to compensate for level differences caused by varying loop resistances to the classrooms.

FIGURE 6

TELERESPONSE: SYSTEM CONFIGURATION

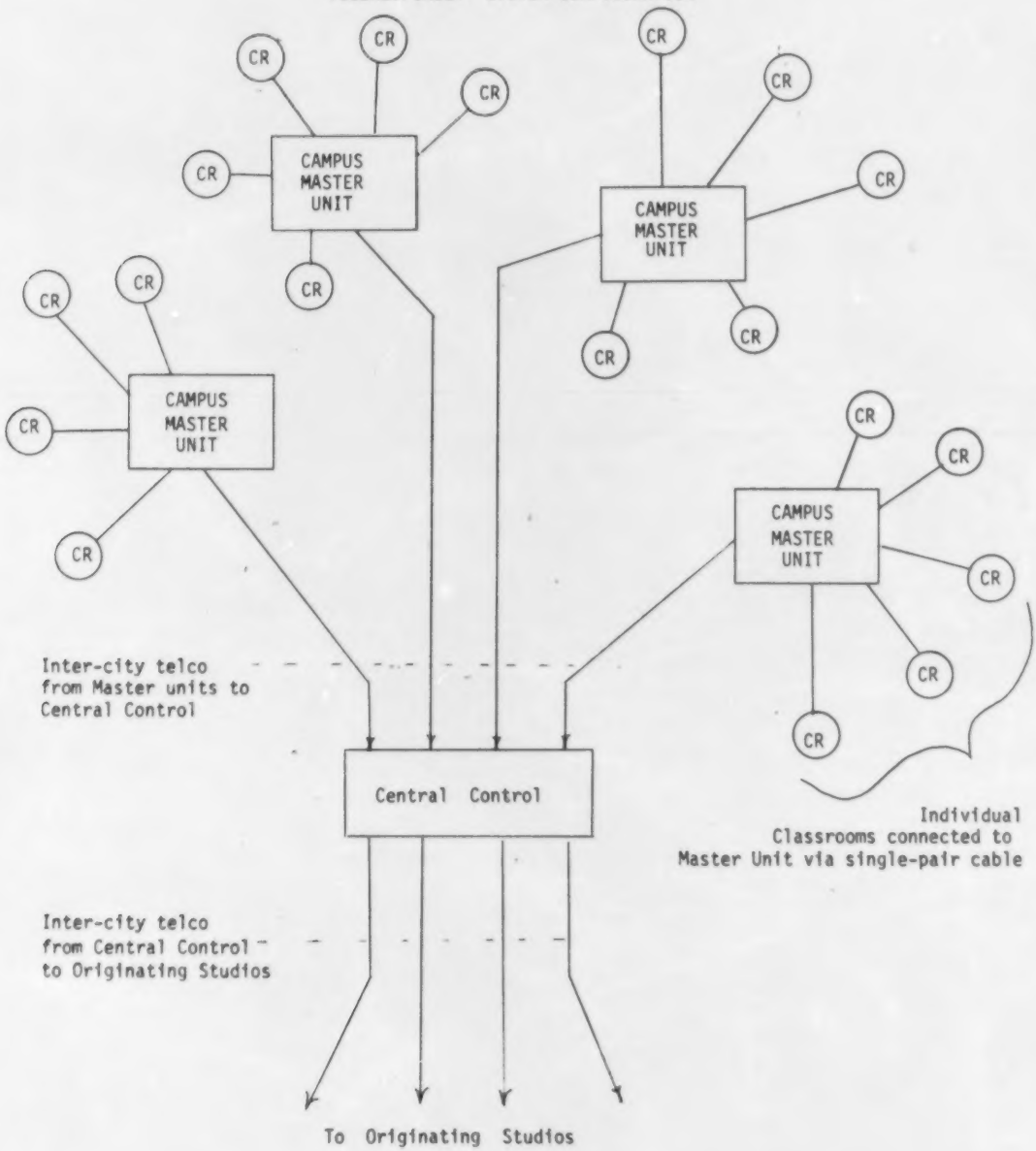
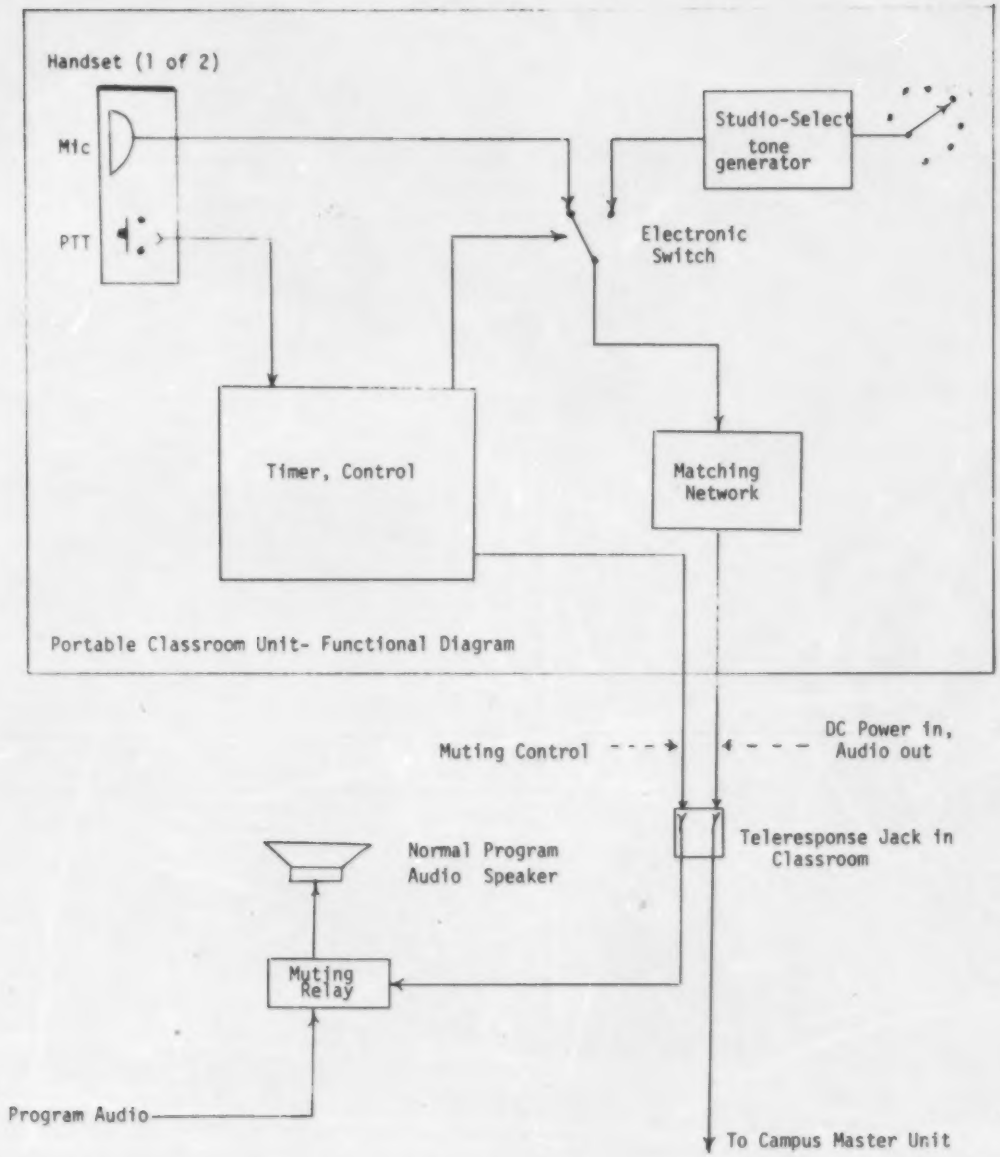

CR
CAMPUS MASTER UNIT
Inter-city telco from Master units to Central Control
Central Control
Individual Classrooms connected to Master Unit via single-pair cable
Inter-city telco from Central Control to Originating Studios
To Originating Studios

FIGURE 7



Central Control

The central control facility links the incoming telephone lines from the various Teleresponse-equipped campuses with similar lines to each of the originating studios. The controller is diagrammed in FIGURE 9. Each incoming line enters an amplifier, which in turn drives one input of the routing switcher and Touch-Tone[tm] decoder. The tone decoders are sequentially scanned by a microprocessor to detect the off-tones and studio select tones from the master units and classroom units respectively. This information is translated into commands for the routing switcher. The actual switch closure is delayed until the end of the studio-select tone so that the tone itself is not sent to the studio.

Hardware/Software Consideration

At our particular installation the use of microprocessor control was dictated in part by the fact that much of the control circuitry for Teleresponse (microprocessor, tone decoders, phone lines) is shared with the Teleprocessing system. The presence of the microprocessor does, however, provide us with a high degree of intelligence in the controller that (1) allows for simpler hardware at other points in the system and (2) allows us easily to expand the system to accommodate special features.

One area of hardware minimization achieved as a result of the microprocessor was the Touch-Tone[tm] decoder modules, which the IHETS staff designed specifically for this project because commercially available

units were bulky, expensive, and not directly compatible with our interface. Our own design uses low-cost phase-locked-loop integrated circuits as tone detectors with no bandpass filtering whatsoever. The elimination of the filters greatly reduces the size, complexity, and cost of our decoders, though it does make them more susceptible to false triggering. Consequently, we employ software "filtering" by requiring a stable decoder output for a fixed number of successive scans before it is accepted as valid. This has proven very effective and has resulted in an overall cost-saving.

Conclusion

Performance of the Teleresponse system has been as expected, and we feel that our design goals were met successfully. Most significant are the reduction of system delay by a factor of 50 (from 15-plus seconds to about 1/3 second) and the improvement of audio quality, in terms of both S/N ratio and elimination of feedback.

FIGURE 8

CAMPUS MASTER UNIT: BLOCK DIAGRAM

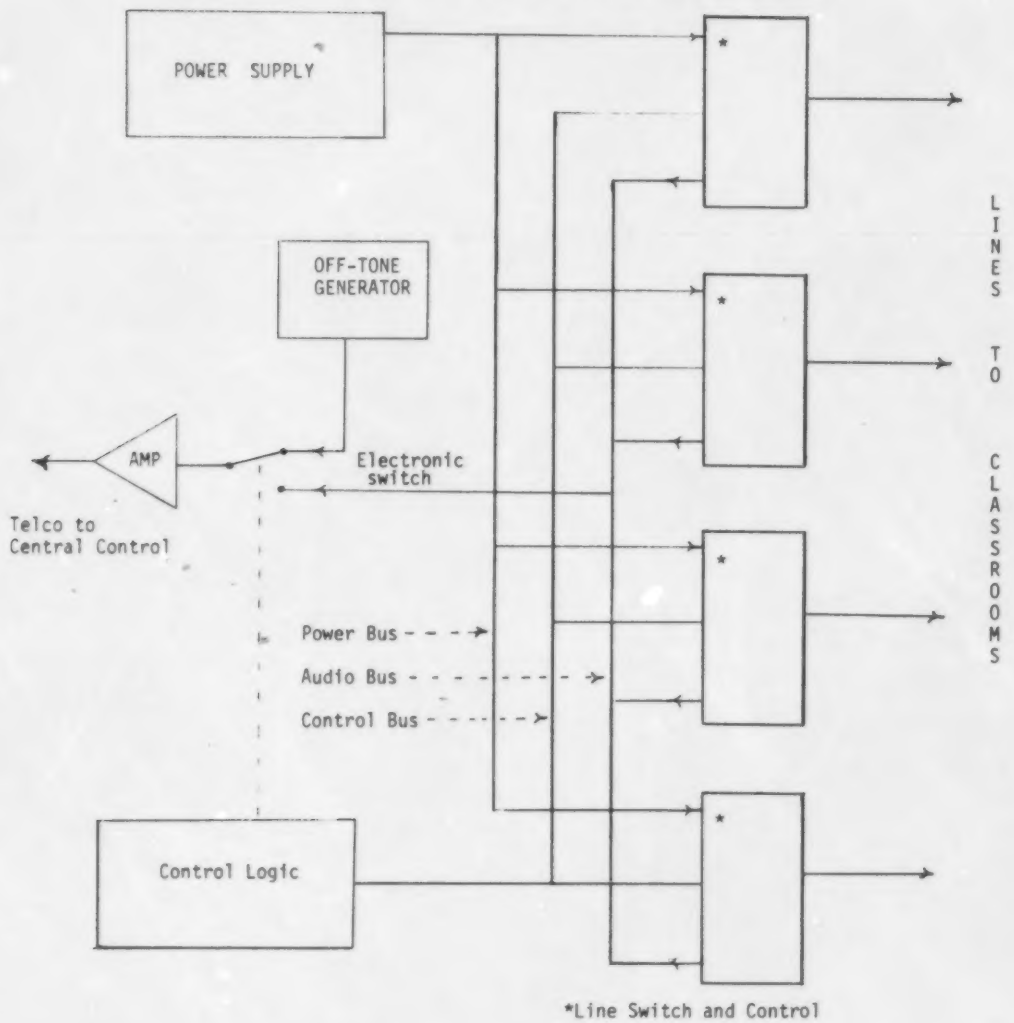
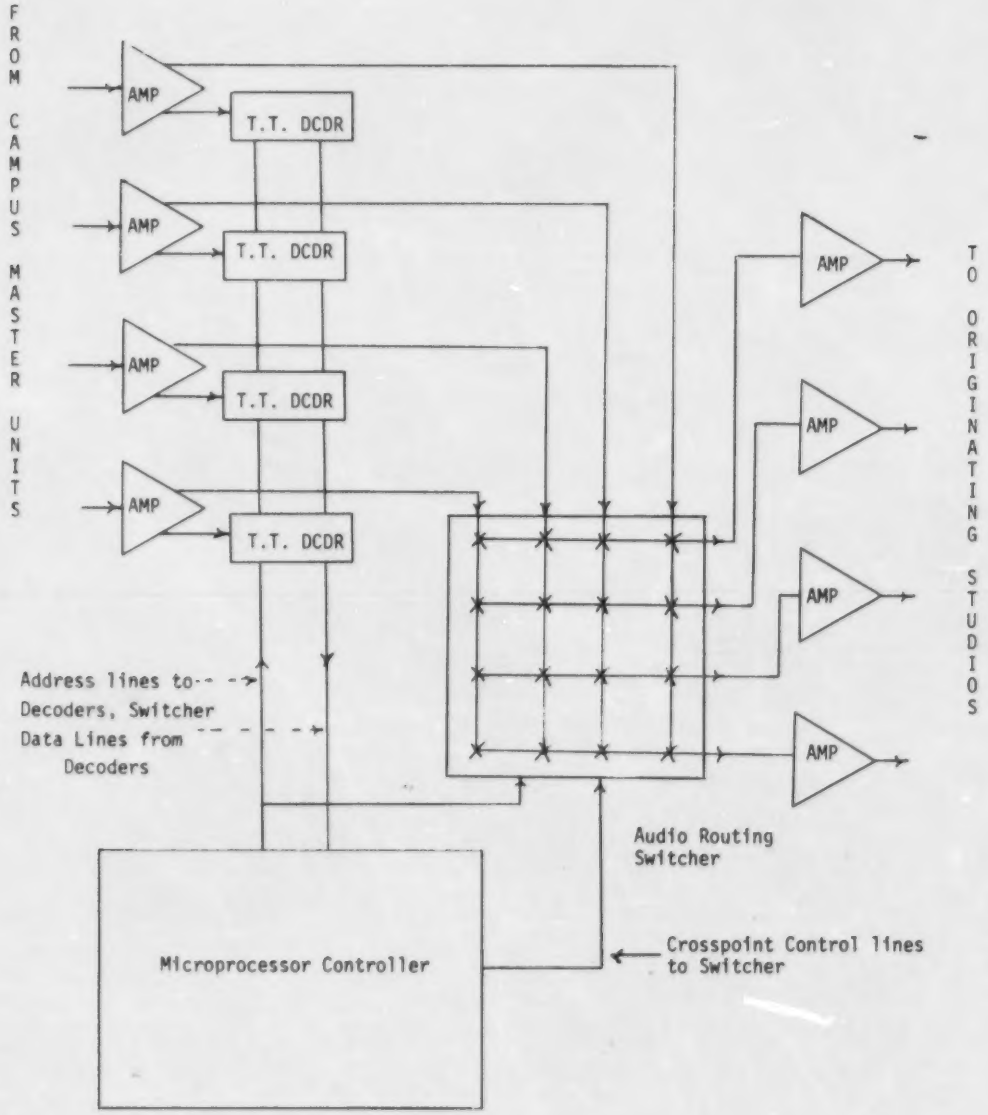


FIGURE 9

CENTRAL CONTROL: BLOCK DIAGRAM



TELECONFERENCING + TELEWRITING = CONTINUING ENGINEERING EDUCATION IN WISCONSIN

Lowell B. Jackson, P.E.
Lorne A. Parker, Ph.D., and
Christine H. Olgren
University of Wisconsin-Extension

The telephone is a look-twice phenomenon. At first glance it appears to be a limited medium with little potential in education. The telephone has thus often been ignored in favor of its more glamorous sisters--television, radio, and computerized instruction. But a second look reveals its particular advantages: the telephone is interactive, flexible, low-cost, widely available, and supported by a well-developed infrastructure. It is also an audio and visual communications channel. Using telewriters, graphic input devices, or slow scan televideo systems, a telephone network lends itself to a variety of instructional formats.

In short, the telephone is uniquely suited to many educational roles; one of the foremost being the delivery of programs to a new generation of adult learners.

Continuing professional education and avocational adult education are two of the fastest growing areas of higher learning. Their rapid growth reflects both a new philosophy of education and the need for professionals to keep abreast of information. It is also the result of the simple demographic fact that the U.S. is becoming an older society. In the present decade, the number of 25 to 34 year olds will increase to 44 percent, and this dramatic shift to an older population will continue well into the 1980's and 1990's.

Higher education has traditionally focused on full time, on-campus youth who enrolled in formal degree programs. However, the new post-secondary student, the adult, has a different view of education and wants learning opportunities to be available in his/her home community. These part-time learners are people of all ages and lifestyles, and they have diverse learning needs. Rather than defining education as a terminal degree program to prepare one for future goals, the adult learner sees education as a self-directed activity that continues throughout life. Education, conceptualized as life-long learning, is a vehicle for ongoing vocational development and leisure-time pursuits.

Aside from a personal interest in professional growth, continuing education is increasingly viewed as a necessity in many fields, including engineering. Practitioners are not only strongly encouraged but often required to enroll in continuing education programs as a practical means of maintaining professional skills amidst rapid changes in technical information. This tendency for professional associations and licensing boards to encourage continuing education will most likely accelerate, forcing educators to revise their curricula and teaching methods.

The adult learner is also most often a distant learner who requires non-traditional delivery systems. To reach adults in their home communities, higher education, which is primarily campus-based, must implement new modes of instruction. However, the development of outreach programs is often limited by budgetary constraints. The delivery of instruction to students at locations near their homes or places of business is usually not justified economically if an instructor must travel great distances to serve a few participants at each classroom site. What alternative, then, is most economically feasible, makes best use of teaching resources, and effectively meets the education needs of distant adult learners?

In response to this question, many educators are turning to the telephone--one of the oldest and most effective mediums. Although this invention is over 100 years old, it remains the basic instrument of the new communications technology. After many small-scale experiments in the 1960's, the telephone's unique advantages are leading to a second generation of educational telephone networks in North America and Europe.

There is almost universal agreement that two-way communication is a desireable and necessary element of distance education[1]. A telephone network is interactive, allowing students and instructor to exchange information, ask questions and receive feedback. The process of interaction between faculty and students and among the students themselves is perhaps the most important of the educational processes. While television and radio may be appropriate for some courses, these essentially one-way systems have disadvantages when discussion and immediate feedback are required.

A television network which uses cables, microwave or broadcast channels, is also costly and cannot easily be modified to incorporate new receiving locations or shift transmit sites. A telephone network can use various combinations of dedicated and dial-up lines to minimize cost while maximizing receive and transmit flexibility. For example, dedicated lines may interconnect any number of remote classrooms while dial-up lines allow people at additional locations to bridge into the network.

The telephone also offers instructional flexibility. Course materials can be modified easily at a reasonable cost. The latest telewriting equipment or slow-scan televideo system can display a variety of graphic or pictorial information to supplement audio instruction.

In Wisconsin the focus has long been on the tele-

phone. The Wisconsin Idea, that the boundaries of the campus are the boundaries of the state, provides the philosophical base for statewide outreach programs. In this supportive environment, the University of Wisconsin-Extension's Statewide Extension Education Network (SEEN) and Educational Telephone Network (ETN) have flourished. Today these telephone networks annually attract over 30,000 students--engineers, teachers, physicians, nurses, librarians, lawyers, businesspeople, social workers, and others[2].

Wisconsin's Statewide Extension Education Network (SEEN)

Wisconsin's Statewide Extension Education Network illustrates the application of the telephone to engineering education. SEEN was developed in 1969 to meet the instructional needs of Extension Engineering. For the past eight years, its unique capability for both audio and visual programming has made it possible to deliver continuing education opportunities to engineers throughout the state, even in the most isolated communities of northern Wisconsin.

SEEN uses leased, commercial telephone lines to simultaneously transmit audio and visual material to a number of widely separated classrooms. Two-way audio communication allows students at any SEEN location to actively participate in the course, asking questions and exchanging information with the instructor and fellow students at all of the connected sites. Electrowriters supply the visual element. Any material that is customarily shown on a classroom chalkboard--equations, diagrams, outlines, graphs, line drawings--can be instantly presented on the electrowriters and transmitted over the telephone lines to all SEEN locations. Students view the material exactly as it is created by the instructor and may comment or ask questions anytime during the session.

The SEEN system links more than 20 locations throughout Wisconsin (see Fig. 1) with approximately 3,500 miles of long-distance telephone lines. These locations were selected according to population densities, educational needs of local residents, and distribution of engineers and other professionals in the state. This configuration allows the network to serve 80 percent of Wisconsin's engineers and technicians. The classroom settings vary. Some are in county courthouses, while others are on campuses or in manufacturing firms. In addition to intra-network communication, a telephone call from anywhere in the U.S. can be tied into the system, providing contact with outside experts.

Programming for Engineering Education

Extension Engineering offers SEEN programs in two general areas: non-credit continuing education and credit undergraduate and graduate education[3].

Continuing education, the largest program area and basis of the entire teleconferencing network in Wisconsin, is directed to the practicing engineer and other professionals. Engineering courses are usually conducted over a six to eight week period, meeting once or twice each week. They tend to be practical "state of the art" courses tailored to the needs of professional engineers. A typical course in the category is the "Industrial and Manufacturing Engineering Refresher" which reviews current techniques in manufacturing processes, scheduling, cost analysis, quality control, and other pertinent topics. Continuing education courses are usually offered between 4:30 and 10:00 p.m. Monday through Thursday, times that are convenient for working engineers. Participants earn Continuing Education Units (CEU) for each program satisfactorily completed.

Wisconsin engineers who enroll in continuing education programs have a unique opportunity to earn a Professional Development Degree in Engineering, a post-baccalaureate degree pioneered at the University of Wisconsin. The P.D. degree program is based on the special needs and personal objectives of the full-time employed engineer. Each student enrolled in the degree program meets with a faculty advisor to establish an organized plan of educational development. The plan is flexible and can be changed to reflect new learning objectives. The P.D. degree requires 120 CEU's. In addition to SEEN courses, units may be earned in short courses, institutes and seminars, workshops, correspondence study, video cassette courses, and individual guided study programs.

Undergraduate and graduate credit courses are available to both off-campus and on-campus students. These courses are customarily conducted during the morning or afternoon hours. Continuing engineering students may also enroll and take the courses at any SEEN site. Resident campus students are usually enrolled in SEEN courses that are simultaneously telecast to the 14 two-year campuses of the University of Wisconsin Center System. This arrangement is often more economical than offering separate classes at each school. Basic courses in statics and dynamics for pre-engineering students, for example, originate at the center in Wausau and are received by about 50 students in groups of four or five at other centers during a mid-morning hour. Although this time of day is not convenient for most continuing education students, several have joined this class. In many cases, the inclusion of working students who are occupationally associated with the subject matter has added much to the relevance and scope of instruction. In such subjects as products liability, welding metallurgy, and pollution control, the experienced engineer has contributed a great deal to the course and challenged the instructor to keep abreast of engineering practice.

Engineering SEEN courses may originate in Madison, Milwaukee, Wausau or other locations, allowing flexibility in program topics and the most effective assignment of faculty specialists from any part of the state.

Administration is a Cooperative Effort

The administration and management of the SEEN engineering programs involves three groups: Instructional Communications Systems, Extension's Department of Engineering and Applied Sciences, and Extension agents at the local level.

Instructional Communications Systems (ICS), a division of University Extension, is responsible for the overall administration of SEEN, coordination of network programming, and management of the technical system. Coordinating functions include program scheduling, instructional design, promotion, and evaluation. ICS also serves as a liaison between Extension Engineering and the local agents. Technical management of the system involves the operation and maintenance of the telephone network, studios, classroom hook-ups, electrowriters, and audio equipment as well as the production, recording and distribution of programs.

In Extension Engineering, course design, faculty identification and all academic matters related to SEEN programs are managed by the director of Electronic Media Programming in Engineering (EMPE). The EMPE director initiates courses, conducts direct mail promotion, and coordinates with other Extension departments, such as math and business and management, concerning offerings

of probable relevance to engineering clientele.

The third branch of administration is provided by the Local Program Administrator (LPA), an Extension agent headquartered at or near a SEEN location. The LPA is the contact person at each site who is responsible for scheduling, managing, and promoting programs at the local level. The LPA may also have a program aide to welcome and assist students, operate the audio-visual equipment, and, in general, help people obtain a positive educational experience.

The Faculty

The instructional services for SEEN courses are supplied by three types of faculty: (1) the professional staff of Extension Engineering: (2) the teaching faculties, primarily in engineering, at 5 or 6 of the 27 campuses in the university system: and (3) the engineering and industrial community with appropriate experience and teaching expertise.

Those in the first category supply the bulk of offerings of the noncredit variety, dealing with subjects of current relevance in their areas of professional competence. These teachers are able to respond quickly to the changing aspirations of continuing engineering clientele. Participation in SEEN teaching is encouraged as a substantial method of professional advancement.

The SEEN courses offered for credit are generally taught by resident campus faculty. Such classes are adapted directly from existing courses offered to students on campus. Although it is not necessary, the courses offered by these faculty members are generally those simulcast to the network from a class of resident students. Teachers in this category have in the past been compensated by an "overload" payment in addition to their regular faculty salary, but they now often engage in SEEN programming as a part of their regular academic activities.

Building upon the long experience of the evening, noncredit offerings based in Milwaukee, some of the SEEN schedule is devoted to simulcast versions of these courses. The instructors may be from any of the three types of faculty. Those not on the staff of UW-Extension are usually compensated at an hourly rate.

Since the network began operation in 1969, more than 100 men and women have served on the SEEN instructional staff.

History of SEEN Engineering Programs

The first SEEN course, a series of 13 sessions on an electrical engineering subject, was convened in 1969 at five separate locations with 19 students. An evaluation by teachers, students, and interested administrators led to planning for expanded use of this concept. During the 1970-71 academic year, a joint program of 16 courses at 12 sites was conducted by joining SEEN with a similar network operated by the University of Illinois. Under this sharing arrangement, resident and continuing education students located in two states enrolled in the courses of their choice through their own university. The first semester (fall of 1970) featured only daytime, credit courses, which were provided primarily for graduate students in civil, electrical, and mechanical engineering, mathematics and computer science, mechanics, and minerals and metallurgical engineering.

During the 1971 spring semester, a number of noncredit refresher-type courses conducted during the evening hours were added. The joint programming effort between SEEN and the University of Illinois ended in the fall of 1971.

Table 1 presents statistics pertaining to the SEEN engineering program throughout its operational life. Since courses differ in length, in credit status, and in design relative to the contact hours required per credit, the Continuing Education Unit has been used to represent the total time spent by the student on the course, including outside reading and study.

Table 1. SEEN Statistics, 1970-1977

Semester	Number CEU	Enrollees	Ave. Enroll/ Course
Fall, 1970	652.8	77	7.0
Spring, 1971	1202.0	157	17.4
Fall, 1971	938.8	76	6.3
Spring, 1972	827.3	88	11.0
Fall, 1972	1324.1	181	15.1
Spring, 1973	936.8	137	15.2
Fall, 1973	906.0	153	17.0
Spring, 1974	1168.6	295	29.5
Fall, 1974	926.1	271	22.6
Spring, 1975	1349.0	400	36.4
Fall, 1975	933.6	183	22.9
Spring, 1976	1051.5	273	24.8
Fall, 1976	735.8	156	26.0
Spring, 1977	733.2	286	28.6
Fall, 1977	1139.4	275	25.0

Experience and expert projection indicate that the engineering portion of the overall SEEN use can be expected, with good management, to maintain an average level of activity per semester characterized by about 1,000 CEU's, generated by about ten courses, each with approximately 25 students enrolled.

The Technical System

A successful educational teleconferencing system, whether made up of two or 200 locations, includes three basic components: (1) the terminal or station equipment; (2) the interconnecting transmission system; (3) the network control center. Many variables influence system design. Included are the number of locations; the geographic area covered; the network configuration, whether the system is a four-wire, multi-point dedicated network or a two-wire dial-up network; and the network's intended use. No two systems are likely to be the same, but several ingredients are common to successful systems:

1. An educational teleconferencing system should be able to connect participants at widely scattered locations.

2. The system should provide a communication's environment that duplicates (as closely as possible) the single-site discussion group.

3. Terminal or station equipment should provide even sound distribution and allow easy participation.

4. The system should be manageable, allowing additional locations and equipment as well as modifications to meet user's changing needs.

5. A network control center should provide a program origination point and essential supervision of the system's technical operation.

6. The transmission system should provide clean, intelligible communication between

all points on the network.

7. The system should be able to connect groups anywhere in the world via regular telephone service.

8. Methods should be developed to quickly recognize and clear any technical problems interferring with service.

9. Equipment should feature the latest technology and be reliable and serviceable.

10. All terminal equipment, station location interfaces, and installation procedures should be standardized throughout the system[4].

With these general capabilities in mind, the three basic components of the SEEN system will be described in greater detail[5].

Station Equipment

Each of the SEEN classrooms has identical equipment: a Darome Convener for two-way voice communication and a Victor Electrowriter* to either transmit or receive graphic material. Using these devices, an instructor can present both audio and visual information to students in a number of widely scattered locations.

The Darome convener is a self-contained, portable conference set that contains microphones and a speaker that plugs into a standard telephone jack and AC power outlet. Students simply press a bar at the base of the microphone to participate in class discussion. Each microphone has a 20-foot cord to allow easy access by students. As many as 16 microphones may be placed in a single classroom using a jack provided on the unit and audio mixers. The audio amplification system is sufficient to provide highly intelligible sound in a classroom seating up to 300 people. The Convener was developed cooperatively by Instructional Communications Systems, Darome Company and Wisconsin Telephone Company to meet the unique need for immediate, two-way communications in an extended geographical area.

Electrowriters permit the transmission, reception and projection of any visual material that is customarily shown on a classroom chalkboard. The instructor writes on the electrowriter transmitter with a special ball-point pen. Pen position and movement create tone signals that are carried over regular telephone lines to electrowriter receivers in SEEN classrooms around the state. These signals are translated by the receiver into an instantaneous and faithful reproduction of the instructor's writing. The image on the receiver is then projected onto a screen, which becomes the classroom chalkboard.

The basic electrowriter system consists of a transmitter and receiver, each of which have an electronic pen, servo-mechanism, and a writing area of 17.5 square inches. The SEEN electrowriters were modified by Instructional Communications Systems to double the writing and viewing area, making, in essence, a two frame system. When one frame is being developed by the instructor, students can see the additional frame which was previously discussed.

*The Electrowriter, manufactured by Victor Comptometer Company, was an early pioneer in visual telephone devices. Recent developments in telewriting and televideo systems are reviewed at the end of this paper.

Distribution Network

The SEEN system uses two dedicated, four-wire, multi-point teleconferencing networks: voice signals are transmitted over one network, while the second network simultaneously transmits the signals from the electrowriters. The combination of two, four-wire networks gives the system greater flexibility, for both audio and visual information can originate from any of the SEEN locations.

SEEN's dedicated systems uses permanently installed facilities leased from the telephone company on a 24-hour-a-day basis. Because the listening centers are permanent, all the points on the system are wired to operate like a "party line". On a four-wire system of dedicated lines, each message is carried on its own pair of wires, eliminating feedback on the return loop. Greater control over signal levels, signal-to-noise ratio, and signal bandpass also results in a transmission quality far exceeding that of a two-wire conference system.

The basic building block of multi-point, dedicated network is a four-wire, six-way bridge. Using this together with existing telephone transmission facilities a network of any size can be designed. These bridges can be located regionally; thus, a network can be formed in a regional building-block fashion. However, the practical size limit for any region is 20 stations because of the noise created by the multiple facilities and circuit terminations. Most of the techniques used to construct a multi-point, private line teleconferencing system are standard operating procedures within the telephone industry. Equipment is available from several manufacturers and suppliers.

An important ingredient in a successful dedicated telephone network is the ability to connect other locations to the four-wire system. By bridging a dial-up call to the private line network, SEEN's coverage is extended to any location where there is a telephone.

Studio Control Complex

The studio control complex consists of studios from which programs may originate and a control room for network operation. It is located on the University of Wisconsin-Madison campus and is operated by Instructional Communications Systems.

The control room and studios were designed to provide smooth, trouble-free programming and to minimize the burden on the instructor. Because all technical aspects of a program are handled by a trained engineer, the instructor is free to concentrate on his/her material and presentation.

The control room is equipped to handle five major functions: program control, telephone network control, studio sound control, monitoring, and network failure detection.

The control room technician uses a console to control program audio sources, such as tape recordings and microphones. For example, the technician loads any insert tapes on the tape machines and plays them on cue from the instructor. The technician also controls the electrowriter signals and records programs for future reference on "make-up" sessions.

Complete control of the entire telephone network is maintained in the control room. Prior to program time, the technician establishes the teleconferencing network, quickly tying-in the appropriate locations. The network of dedicated lines can be arranged as programs and en-

rollments dictate. Participating locations can be selected for total system programming, or regional networks can be established and programmed simultaneously. Dial-up bridging controls also allow the technician to tie regular calls into the private network.

Studio sound system equipment is used to control the distribution of audio programming in three areas: in the studios so the instructor can hear the response from the network, in the control room so all audio channels can be monitored, and to administrative offices so staff can listen to programs. General building paging with emergency override to all areas is also provided by the control room.

Monitoring is an important function that contributes to the overall quality of network programming. Console controls and VU meters allow the technician to cue and listen to any of the audio sources without connecting them into the network or program. The technician is also able to selectively monitor the output levels at the distribution console and correct the level quickly, if necessary. Another monitoring feature is used to check if transmit signals are getting through the master bridges at the telephone company. The quality of the visual presentation is monitored on an electrowriter receiver in the control room.

Quick detection and resolution of network failures is vital to the success of any teleconferencing network. Within a 16-hour programming day, an average of three or four system failures may occur on Wisconsin's two educational telephone networks. Adequate monitoring and test features are therefore necessary to minimize network downtime and program interruption. An experienced control engineer can isolate a problem quickly and identify the source--station equipment, transmission line, carrier system, or some other characteristic failure. For more complex failures, the problem can be reported to the telephone test board which then helps to isolate and correct it. Given the ability to quickly isolate a region, the technician can bypass the problem area and continue to operate the remaining parts of the network. A control room microphone allows the technician to talk to any point on the network.

A separate auxiliary SEEN control room serves as a back-up unit. Although not as sophisticated as the main control room, the auxiliary equipment is capable of transmitting a SEEN program to all network locations.

The SEEN studio is adjacent to the auxiliary control room, separated by glass windows. The studio is equipped with an electrowriter transmitter, microphones, projection screen, and table and chairs.

The electrowriter transmitter is contained in a special table that allows it to be flush with the table surface for ease of writing. The electrowriter also sits on a swivel base that can be adjusted for the most comfortable position. A microphone head-set frees the instructor's hands for writing.

Eight bar-activated, table microphones allow students to interact with participants at other SEEN locations. Tables and chairs are arranged as a classroom setting, with each table having two microphones.

An electrowriter receiver in the auxiliary control room is used to project the visual presentation onto the screen in the studio as the instructor writes on the transmitter.

Program Planning

Effective use of the SEEN network depends on careful program planning. From the inception of a topic through final production, programming is a complex process that requires close coordination among the instructional department, ICS staff, and local Extension agents.

Program planning has evolved from direct experience with educational teleconferencing in Wisconsin. As new approaches proved successful, they were incorporated into a set of programming guidelines that have become standard operating procedures. These procedures delineate the steps in the programming process and specify coordinating functions, but they are flexible enough to accommodate diverse program needs.

Program planning basically involves seven stages: content selection and development, network scheduling, operational design, program announcement, promotion, registration, and production[6]. Standard procedures guide programmers through this process. A 20-minute video tape acquaints programmers with the many elements involved in effective programming, including support services available at ICS and other Extension departments. A brochure on planning procedures is also available to programmers.

Content Selection and Development

The content of SEEN engineering courses is initiated and developed by Extension Engineering faculty. The EMPE director oversees the selection of courses and instructors.

For continuing education courses, engineering has found that effective programming is based on an accurate determination of the educational needs of its clients. This is accomplished in several ways: questionnaire surveys; voluntary suggestions from both instructors and students; advice from persons such as industrial training directors or those serving on various continuing engineering studies advisory committees; and spontaneous "targets of opportunity" that indicate a subject likely to be of current interest to a specific clientele. For example, close watch is kept on legislation which appears to require an increase in some technical skill or knowledge, such as a new uniform single-family housing code or a solid waste disposal law.

Graduate and undergraduate courses are based on the more traditional requirements of a degree program as well as the willingness of an instructor to offer a course in the SEEN mode.

Scheduling of Programs

Program time on the SEEN network is usually scheduled at an annual scheduling meeting held in January. One or two weeks before the meeting, programmers are invited to submit programming requests for the academic year. Using established procedures, these requests are sorted and preplotted on a master schedule to best utilize network hours and resolve conflicts before the meeting. The meeting itself gives programmers an opportunity to review the entire schedule, negotiate hours if needed, and firm up their program times. Two weeks after the meeting, all requested network time is rechecked and confirmations are sent to the programming departments. Requests for time can also be made during the year, but it is to the advantage of the programming department to reserve network time at the annual scheduling meeting.

Operational Design

The overall operational design of a program involves both the programming department and ICS. Each group has certain responsibilities but work closely in planning the following:

- TARGET CLIENTELE: Identifying target clientele and program objectives
- BUDGET: Determining program materials, mailings, production, support staff, honorariums for speakers, people needed to make the program successful and arriving at a program fee
- PROMOTION AND PUBLICITY: Planning brochures, press releases, radio spots, feature stories, radio-TV appearances
- PRODUCTION TIMETABLE: Establishing prerecording, telephone tie-ins, remote program origination, A-V materials, cassette packaging
- INSTRUCTIONAL APPROVAL: Obtaining department chairman and divisional dean's approval
- REGISTRATION PROCESS INVOLVED
- EVALUATION INSTRUMENT TO BE USED
- PROGRAM MATERIALS: Determining handouts, books required
- COURSE FORMAT TO BE FOLLOWED
- STATION SELECTION: Serving statewide or certain locations
- PROGRAM INITIATION REQUEST: Confirming the operational design; also the official contract between the programming department and ICS

Program Announcement

A program announcement is prepared by the ICS coordinator in conjunction with the program department to provide the local program administrator (LPA) and network coordinators with pertinent information about the program. A schedule card is enclosed which must be returned within a given time, usually two weeks, if the network facilities cannot be scheduled. The LPA determines location availability, and uses the schedule card to inform ICS accordingly. From the schedule cards returned, a listing of confirmed locations is sent to the program department and the registration office which keeps a checklist for location requests. The program department then finalizes its program brochure identifying specific network locations. The program announcement, with its approximate 60-day lead time, gives the LPA's and the programmer ample time to properly schedule and promote the programs.

Promotion

The promotional strategy for SEEN engineering courses has been largely based on direct-mail announcements. Approximately 20,000 persons in Wisconsin and some engineers in adjacent areas in Iowa, Minnesota, and Illinois receive brochures. An internal mailing list is maintained by the department, and a few outside lists are also used.

There are 3 types of direct-mailings: (1) an annual directory that lists all Engineering Extension offerings and describes the SEEN system along with other instructional modes; (2) a catalog of EMPE courses (SEEN and various television formats), either on a semester or annual basis; and (3) monthly-reminder brochures of upcoming programs, generally grouped in interest categories. The brochures are sent for special courses and serve as a reminder to those who have received the general announcement, particularly when a course does not begin until mid-semester.

Use is also made of more general promotional vehicles, such as the newspaper tabloid prepared each semester by ICS and the Program Information Office. It lists all programs by department and includes dates, time, fee, clientele, and a general description of the course. The tabloid is distributed by the local program administrators and has proven to be an economical and convenient promotional tool. Newspaper ads and public service TV spots have also been employed on occasion.

Of growing importance to promotion are the local program administrators. Their local contacts with potential clientele are increasingly effective in generating enrollments, and their efforts are enhanced by special news releases prepared by Extension's Program Information Office for distribution by the LPA in local news media.

Registration

Registration for SEEN engineering courses is primarily by mail in advance of the first program. Enrollment forms provided on program brochures are addressed to the EMPE director. After listing the registrants for each course, the director forwards the enrollments to Extension's Central Registration Office, which processes the forms and deposits the enrollment fees in the programming department's account. Copies of the registration forms are sent to the registrant, the programming department, and to the local program administrator at the participant's location.

Production

Prior to the first program session, ICS staff contact the instructor to confirm the production format for the course. If an instructor is unfamiliar with network facilities and production capabilities, an appointment is arranged to demonstrate the equipment and explain the production process.

Production requirements for each program are written-up on a standard form and given to the network engineer for use at transmission time. This includes the locations participating in the program, origination site, tape inserts, dial-up telephone participants, and tape duplication services.

Programs on the SEEN system can originate at any of the network locations. However, most engineering courses originate in Madison,Milwaukee or Platteville where the majority of faculty members are located. The programs may be produced live or pre-recorded for later telecast. During all programs the audio and visual elements are monitored by the control room engineer to transmit the highest quality program possible.

In addition to the SEEN studio and regular monitor-control functions, other production services are available to faculty, including:

- FACULTY PRODUCTION LAB: A workshop area where faculty may use cassette and reel tape machines, turntable and telephone for easy and convenient recording, editing and listening to program materials.
- U-TAPE-IT STUDIO: A self-operated, push-button control studio in which faculty record lecturers and interviews.
- STUDIO RECORDING AND EDITING: A technician-controlled production service available for single-voice and multiple-voice recordings, such as for role playing and panel discussions. Editing assistance is also available.
- TAPE DUPLICATION: Cassette and reel copies of programs available for purchase by faculty and students.

Instructional Design

An interactive telephone network such as SEEN is a

unique instructional mode. Although it is similar in some ways to face-to-face teaching, there are important differences. Effective programming, therefore, requires that certain elements be incorporated into course content and teaching style.

The design techniques used in SEEN programs are based on 12 years of experience with Wisconsin's educational telephone networks and accumulated research in communications, adult education, listening, and learning theory. Workshops, printed materials, and faculty consultations help SEEN instructors implement these techniques and use the network most effectively.

Four design elements are considered essential to interactive programs: personalizing the experience, varying the style of presentation, seeking participation from the statewide audience, and obtaining feedback [7,8]. Each design element is incorporated into a program by using a number of practical techniques.

Personalizing the experience helps students feel comfortable in the distant learning environment by creating a congenial atmosphere and group rapport. SEEN instructors usually adopt an informal teaching manner, allowing their personalities to come through. Frequent use of names and locations identifies participants. By emphasizing common objectives and the sharing of ideas they are part of a group.

Style of presentation involves many elements, all of which contribute to the goal of helping the learner understand and remember the material. By coordinating the interplay between audio and visual information, the instructor guides the students through the learning process. A variety of illustrations, such as drawings and graphs, presented on the electrowriter amplify ideas and underscore key points. The visual development of equations accompanied by a clear and concise verbal explanation help the learner understand the mathmatical relationships. Instructors often exploit the two-frame capacity of the electrowriter by reviewing the material previously presented, which contributes to the clarification and retention of ideas. Topic outlines, bibliographies, lists of resources available for self-study, and other hand-out materials distributed before the program help students organize the learning experience.

Participation is an integral part of media like SEEN. However, it does not occur automatically. Detailed class rosters, for example, are used to direct questions to those with specific engineering experience and training. At the beginning of a program, many instructors converse informally with students at different locations to "break the ice" and stimulate discussion.

Feedback enables both the student and the teacher to assess the learning experience, achieve program objectives, and improve over-all performance. Informal question-and-answer periods interspersed throughout the program provide immediate feedback on how well the material is understood. Many instructors ask participants to send in questions and comments, others make periodic phone calls to students in their classes to find out if the program is meeting objectives. Individual or small group projects and written responses to specific questions are other means of obtaining feedback.

Evaluation

Evaluation is an integral part of the programming process that contributes to the overall effectiveness of the SEEN system and the quality of individual courses. Three types of evaluations are conducted for SEEN engineering programs: (1) system surveys of SEEN clientele at three-year intervals; (2) surveys directed to the participants in a particular SEEN course; and (3) spot evaluations to facilitate the critique and management of certain courses.

System Evaluation

The purpose of the system evaluation is to determine the strengths and weaknesses of current programs so that improvements can be made in future program design. The evaluation results show the students' assessments of both the instructional and technical elements of the system. These results are analyzed by the program coordinators, educational specialists, and individual instructors; and recommendations are incorporated into the design of subsequent programs. The system evaluation also provides peripheral benefits. For example, it functions as an additional channel of communication between the participants and the instructor, and it is a tangible sign of interest in student opinion[9].

System evaluation of the SEEN engineering programs were conducted in 1974 and 1977. Survey questionnaires were mailed directly to SEEN engineering clientele. Response was voluntary, resulting in a sample size of 282 respondents in 1974 and 244 in 1977.

The first part of the evaluation instrument included a series of items that surveyed the demographic characteristics of the SEEN engineering students. Factors such as age group, participation in various forms of continuing education, extent employers encouraged education, and distance traveled to SEEN courses provide a demographic profile of the SEEN student that is useful in identifying future participants and in structuring programs for this particular group.

Some of the student demographic data is presented in Table 2. The percentages for the two surveys show little difference in any category except the degree of participation in SEEN courses. The number of respondents who are regular SEEN customers, as defined by enrollment in more than one such offering, is more than twice as great in 1977 than in 1974. This indicates the establishment of a special SEEN constituency.

A statistic which bears upon the justification for off-campus network programming is the distance traveled one-way by the student to the SEEN classroom site. Approximately two-thirds of the participants were within ten miles of course access both years. An item added to the 1977 survey revealed that nearly two-thirds of the students were located more than thirty miles from a university campus with any technical curriculum and one-third were more than one-hundred miles away.

The acceptance of SEEN as a delivery system is indicated by the percentage of students who said they would participate in another course. In both years seven out of eight students responded affirmatively.

The second part of the evaluation systematically measured student response to the engineering programs. It contained thirty descriptive statements related to the instructional and technical aspects of the courses. These items were classified into six general categories:

1. The functioning of the technical components of the programs, such as the audio network and electrowriter.
2. The lecturer's delivery of course material.
3. The course content and other program materials.
4. The arrangement of the physical facilities at the SEEN classroom site.
5. The program's organization and involvement of

Table 2. Selected Demographics of SEEN Engineering Students

	1974					1977				
Number of Employees in Company	0-10 4%	11-50 16%	51-200 20%	200+ 59%		0-10 7%	11-50 20%	51-200 16%	200+ 55%	
Age of Respondent	20-30 31%	31-45 42%	46-60 24%	60+ 1%		20-30 25%	31-45 48%	46-60 26%	60+ 1%	
Percent of Tuition Refund by Employer	0-20 15%	20-40 2%	40-60 15%	60-80 11%	80-100 49%	0-20 16%	20-40 2%	40-60 14%	60-80 16%	80-100 46%
Distance Traveled One-way to SEEN Site (miles)	0-2 14%	3-10 49%	11-30 27%	30+ 7%		0-2 18%	3-10 48%	11-30 24%	30+ 7%	
Have Participated in More Than One SEEN Course			19%					42%		
Would Participate Again in a SEEN Course			86%					83%		
Would Not Participate Again in a SEEN Course			10%					9%		

the participants in discussion.

6. The overall effectiveness of the system.

Each of the first five categories corresponded to a primary system objective and was evaluated by at least five descriptive items on the survey. One statement at the end of the evaluation asked for the participants' overall feelings toward the program. Responses to each statement were marked as a numerical rating along an ordinary scale as follows: 1 - strongly disagree (SD); 2 - disagree (D); 3 - neutral (N); 4 - agree (A); and 5 - strongly agree (SA). The data collected were analyzed statistically to derive the mean, percentile, and standard deviation for each category and the individual items. Their reliability, quality, and predictive power were measured by a MERMAC computer program.

The respondent's evaluation of the engineering program is summarized in Table 3. There is little difference in the mean values of each category for the two years. The data collected in 1977 shows a slightly higher level of satisfaction with the instruction (lecturer, material, organization) and slightly lower rating of the physical aspects of the system.

Table 3. Evaluation of SEEN Engineering Programs

1974

Category	Mean	Reliability
Technical	3.78	.77
Lecturer	3.58	.84
Material Presented	3.59	.70
Facilities	4.12	.58
Organization	3.55	.80
Total, All Items	3.73	.92
Overall Effectiveness	3.69	---

1977

Category	Mean	Reliability
Technical	3.65	.74
Lecturer	3.67	.80
Material Presented	3.69	.84
Facilities	3.80	.62
Organization	3.66	.77
Total, All Items	3.69	.93
Overall Effectiveness	3.73	---

A typical individual statement in the survey was, "The mechanical quality (line noise, etc.) of SEEN audio reception was satisfactory." The response to that particular item, which is in the category of technical components, was:

SD	D	N	A	S	Omit	Mean	Standard Deviation
2%	7%	17%	50%	18%	6%	3.79	.93

Individual Course Surveys

Whenever the enrollment in a SEEN engineering course exceeds 55 students, a survey somewhat different from the system evaluation is used. This might occur once or twice during a year's programming. The results attained are strikingly similar to the system survey results. The value of such individual evaluation in the critique and adjustment of the course instruction is substantial. For the individual course survey a number of questions peculiar to the subject are included.

Table 4 shows the summary results of a 40-question survey for a new course which was designed to be repeated once each year and features a faculty of six to eight speakers. The survey was developed specifically for that course. Items scoring below 3.5 are candidates for refinement.

Table 4. Summary of Single Course Survey

Category	Mean	Reliability
Technical	4.16	.48
Lecturer	3.74	.67
Organization	3.52	.62
Knowledge	3.10	.71
Competency	3.13	.76
Content	3.67	.82
Total Overall	3.59	.92

Spot Evaluation

In a number of situations where the size of a class is so small as to reduce the statistical significance of the response, a standard evaluation form is distributed which facilitates critique and management. The return in this category may be computer-processed, as are the other types discussed above, but are more likely to be "hand-processed". These spot evaluations are particularly valuable in connection with first-run courses with fresh instructors, new topical material, or where attendees appear to be distinctly different than the ordinary class. Often these evaluations are deployed in response to random participant feedback which calls for substantiation. Many of the instructors routinely distribute and collect these responses for their own use in refining the presentation or material content of a course.

Costs

The costs of the SEEN system include those for administration, promotion, printed handout materials, instructional fees, line costs, engineering equipment maintainance, and a variety of support services such as those involved in conducting surveys, faculty training, and local site coordination.

The university costs of the SEEN engineering program in 1977 totaled approximately $60,000, with about half being spent on instructional costs and half on network operation. Income from student tuition amounted to some $30,000. Thus, the current self-support level is 50 percent. It is anticipated that continually improved program management and market development might cause this support level to reach 60 percent during the next several years.

Costs in higher education are usually evaluated by two approaches: (1) cost per student contact hour, and (2) cost per continuing education unit. Using the first approach, the SEEN engineering program cost about $17 per student contact hour. In 1977 some 550 students were enrolled in courses with 460 instructional hours at a total university cost of $60,000 (550 x 460 = 253,000 student contact hours ÷ $60,000 = $17). Using the total number of 18,072 CEU's earned in SEEN engineering courses, the cost is about $33 per CEU. A comparison between the cost per CEU of SEEN instruction and other delivery formats is shown in Table 5.

Another approach to the presentation of the various cost data is to relate student and university costs to the percent of "self-study" required of the student to participate in a particular type of instruction. Figure 2 clearly indicates a decrease in cost as the self-study

Table 5

Comparison of Continuing Engineering Education Formats, 1977

Format	Methods used	Clientele served	Average annual # of students	Level and scope	University cost: dollar/student CEU	Student cost: dollar/student CEU
Correspondence study (worldwide)	Self-study with study guides and some audiovisual aids	Students remote from campus	1,800	Mostly undergraduate. Usually general interest topics. Seldom enough demand to justify advanced specialized topics.	$14.00	$8.25
Institutes and short courses (Madison & Milwaukee)	Lectures and workshops	Practicing engineers and technical management	12,000	Very applied. Generally state-of-the-art	$85.00 (institute) $75.00 (short course)	$100.00 $75.00
SEEN (Statewide)	Electrowriter network	Very broad range	550	Applicable to wide range from college courses to single-topic state-of-the-art	$33.20	$20.00
Video cassette courses (worldwide)	Study Guides, Video Lessons	Students & Practitioners on & off campus	1,000	Same as SEEN No interaction	$16.00	$16.00
Evening classes (Milwaukee)	Lectures and labs	Working students, sometimes without company financial support	775	Often similar to college courses. Usually more applied. Some state-of-the-art programs	$35.00	$30.00
University campus courses (Madison)	Lectures and labs	Resident students, part-time students employed locally	2,600	Generally aimed at resident campus students - continuing engineering students can enroll	$22.50 (undergraduate) $41.25 (graduate)	$5.50 $12.25

component increases. The figure also shows the effect of management to bring student and university costs together, whereas the on-campus difference for a public instruction is quite pronounced.

There are additional student benefits which are difficult to estimate--those cost savings for travel between business and campus as opposed to SEEN locations in their home community. The surveys in 1974 and 1977 showed that as in the case of other formats such as Institutes and Short Courses, a great deal of the continuing engineering students' direct expenses are often refunded by his or her employer.

The Future of the Telephone in Education

The telephone has served Wisconsin well in extending educational opportunities to engineers and others throughout the state. The teleconferencing networks have increasingly grown in number of enrollments, programming hours, and variety of classes offered. This expansion is expected to continue as more and more adults seek off-campus educational programs for professional development and personal growth.

Supporting the Wisconsin experience are many studies that show the telephone to be an effective, inexpensive educational medium. Rao and Hicks [10], for example, reviewed 18 experiences in teaching by telephone and concluded that students learn as much or more in telephone classes as in face-to-face discussion. Similar conclusions were drawn by Hoyt and Frey[11], Pellet[12], Blackwood and Trent[13], and many others.

The future of the telephone in education depends not only on its effectiveness as an audio medium but a also on its capacity to provide quality visual information. This consideration is especially important in technical fields like engineering where instructors often present mathematical and graphic material. New telewriting and televideo systems have recently been introduced that have great potential in education. A variety of graphic and pictorial information can be presented over regular telephone lines using improved electro-mechanical pens, graphic tablets, video writers, electronic blackboards, computer-graphics systems, or slow scan televideo systems. Some of these devices and their manufacturers are summarized in Table 6.

Electro-mechanical pens that write on paper-covered surfaces, such as the Victor Electrowriter, are probably the oldest of the telewriting systems, entering the market of telephone technologies in the early 1960's. Since then many improvements have been made to increase the resolution and accuracy. Their relatively low cost also makes these devices attractive for transmitting hand-drawn graphics--equations, line drawings, outlines, diagrams, graphs.

Graphics tablets, video writers, and the electronic blackboard are similar in a number of ways. Hand-generated material is accurately reproduced by writing directly onto conductive surfaces. Graphics tablets usually have electronic grids or specially coated transparent sheets that sense pen movements. To produce graphic material on a video writer, one writes directly onto a TV monitor with a light pen. Ordinary chalk is used to present information on the pressure-sensitive electronic blackboard made by Bell Laboratories. The Electronic Blackboard has been tested in continuing engineering education for a number of years at the University of Illinois at Champaign-Urbana. All of these systems digitize the information and display it on monitors at remote locations.

Computer-graphics systems are capable of showing both written material and computer-generated graphics. A graphics tablet is used for hand-drawn information. In addition, these systems are programmed to perform various computer graphics, such as constructing bar charts and diagrams. Some have programmed symbols that can be instantly placed at any point on the televised display. These symbols may be mathematical or simple figures that can be animated to show, for instance, an airplane flying through a cloud or a person walking. Many firms have lines of computer graphics systems or will design packages especially to meet the user's needs[14].

Slow scan televideo systems add another dimension to telephone instruction--the capacity to present pictorial information as well as graphics. Any image that can be captured by a video camera can be shown on a televideo system, including views of the instructor and classroom, outdoor scenes, written and prepared material. The picture is transmitted over the telephone lines during a number of seconds and shown as a frozen image on a remote TV monitor. Past systems were quite limited, used mainly for security-surveillance operations. However, the recent application of electronic technology has greatly improved their performance and versatility.

The new telewriting and televideo systems provide additional flexibility for visual instruction over a telephone network. Combined with its proven effectiveness as an interactive audio medium, the telephone can play a unique role in education--a role that extends teaching resources to adult learners in many distant locations.

Table 6. Selected Telewriting and Televideo Systems

Capability	Device	Manufacturer
Hand-drawn graphics	Electro-mechanical pens	
	Telescriber	Telautograph Corporation
	Telenote/Telescreen	Talos Systems, Inc.
	Electrowriter	Infolink Corporation
	Graphic tablets	
	Audiographic Systems	Interand Corporation
	Graphic Tablets	Tektronix, Inc.
	Intelligent Digitizer	Summagraphics
	Cybergraphic Systems	Talos Systems, Inc.
	Video writer	FOR-A Company, Ltd.
	Electronic blackboard	Bell Laboratories
Hand-drawn graphics and computer-generated graphics	Computer graphics systems	
	AD/380 Design System	Auto-trol Corporation
	Interactive Graphics Systems	Tektronix, Inc.
Hand-drawn graphics and video pictures	Slow scan televideo systems	
	Phone Line Television Systems	Robot Research, Inc.
	Narrowband Video Systems	Colorado Video, Inc.
	Telephone Video Systems (monochrome and color)	NEC-America, Inc.

REFERENCES

1. Holmberg, B., Distance Education: A Survey And Bibliography, London: Kogan Page, Ltd., 1977.

2. Parker, L. A., "Teleconferencing As An Educational Medium: A Ten Year Perspective from the University of Wisconsin-Extension," The Status of the Telephone in Education, University of Wisconsin-Extension, Madison, Wisconsin, 1976.

3. Klus, J. P., and others. Continuing Education for Engineers: A Case Study of the University of Wisconsin-Extension Engineering Department. UNESCO International Working Group on Continuing Education of Engineers, University of Wisconsin-Extension, Madison, Wisconsin, 1974.

4. Parker, L. A., M. A. Baird, and D. A. Gilbertson, "Introduction to Teleconferencing," The Telephone in Education (Book II), University of Wisconsin-Extension, Madison, Wisconsin, 1977.

5. Braun, D., D. A. Gilbertson, and H. C. Hansen, "ETN: A Technical System," The Status of the Telephone in Education, University of Wisconsin-Extension, Madison, Wisconsin, 1976.

6. Seaman, M., "Operational Aspects for Program Planning," The Status of the Telephone in Education, University of Wisconsin-Extension, Madison, Wisconsin, 1976.

7. Baird, M. A., "Designing Teleconferencing Programs: Some Clues from the Wisconsin Experience," The Status of the Telephone in Education, University of Wisconsin-Extension, Madison, Wisconsin, 1976.

8. Monson, M., "A Report Investigating Teaching Techniques Used Over the Educational Telephone Network at the University of Wisconsin-Extension," A Design for Interactive Audio, University of Wisconsin-Extension, Madison, Wisconsin, 1977.

9. Parker, L. A., "Teleconferencing As An Educational Medium: A Ten Year Perspective from the University of Wisconsin-Extension," The Status of the Telephone in Education, University of Wisconsin-Extension, Madison, Wisconsin, 1976.

10. Rao, P. V., and B. L. Hicks, "Telephone Based Instructional Systems," Audio-Visual Instruction, 4, 18, 1972.

11. Hoyt, D. P., and D. W. Frye, The Effectiveness of Telecommunications as an Educational Delivery System. Kansas State University School of Education, Manhattan, Kansas, June, 1972.

12. Pellet, V. L., The Comparative Effectiveness of the Educational Telephone Network and Face-to-Face Lecturers for University Extension In-service Training. Unpublished Doctoral Dissertation, the University of Wisconsin, Madison, Wisconsin, 1970.

13. Blackwood, H. and C. Trent. A Comparison of Face-to-Face and Remote Teaching in Communicating Educational Information to Adults. Kansas State University, Cooperative Extension Service, Manhattan, Kansas, October, 1968.

14. Machover, C., M. Neighbors, and C. Stuart, "Grapics Displays: Factors in Systems Design," IEEE Spectrum, Vol. 14, No. 10, Oct. 1977, pp. 23-27.

STATEWIDE
EXTENSION
EDUCATION
NETWORK
1976-77
Superior
Rhinelander
Rice Lake
Medford
Menomonie
Wausau
Marinette
Eau Claire
Green Bay
Marshfield
Menasha
Manitowoc
LaCrosse
Fond du Lac
Sheboygan
Richland Center
Baraboo
West Bend
Madison
Waukesha
Milwaukee
Platteville
Kenosha
Janesville
UNIVERSITY OF WISCONSIN-EXTENSION
INSTRUCTIONAL COMMUNICATIONS SYSTEMS

Figure 1

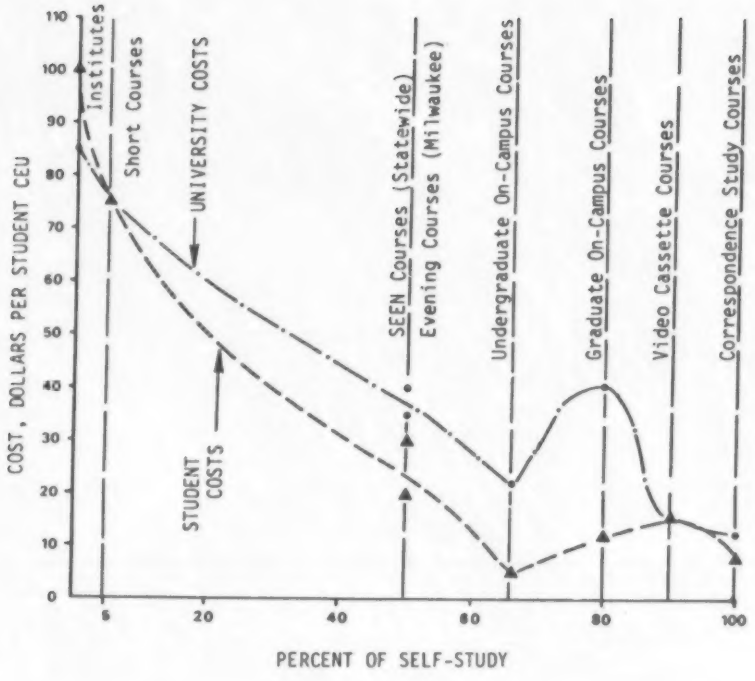

COST, DOLLARS PER STUDENT CEU
110
100
90
80
70
60
50
40
30
20
10
0
Institutes
Short Courses
UNIVERSITY COSTS
STUDENT COSTS
SEEN Courses (Statewide)
Evening Courses (Milwaukee)
Undergraduate On-Campus Courses
Graduate On-Campus Courses
Video Cassette Courses
Correspondence Study Courses
5
20
40
60
80
100
PERCENT OF SELF-STUDY

Figure 2

172